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DETERMINATION OF COMPLEX ACOUSTIC IMPEDANCE AND METHODS OF ACOUSTIC SPECTRAL ANALYSIS

John K. Domen

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13. ABSTRACT

To expedite the efficient analysis for non-lethal acoustic data, a compilation of digital signal techniques was formulated and exercised on data being received from acoustic sources and experiments. This compilation with examples is presented as an initial guide to the practical extraction of material acoustic impedance and of signal frequency and energy distribution, for eventual use as inputs for finite element consideration of material-acoustic interaction.

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OBJECTIVE

During the analysis of experimental acoustic data from the non-lethal program, a series of technical procedures was compiled to assist in a consistent and standard method of data interpretation. The addressed procedures in signal analysis represent a main core of these methods, and are given with a representative example with software programs (FORTRAN) to expedite the handling of data. Of specific concern was the extraction of material acoustic impedance, and a sequence based on literature is given with an example to facilitate this extraction, for use in finite element programs for acoustic interaction with material.

DETERMINATION OF COMPLEX ACOUSTIC IMPEDANCE

Though many references discuss the mathematical and experimental determination of complex acoustic impedance, the approach used by three references was selected. Their method uses the common procedure of determining material impedance by means of a single microphone in an impedance tube (refs 1 through 3).

A common numerical example is used for all three references, and English units are used [feet, seconds]. The first two methods are equivalent, though the mathematics is expressed differently. The third accounts for some attenuation of the signal down the tube.

FORTRAN programs are provided in tables 1 through 3, with an example and input and output files for the listed program. These methods use a sequence of complex quantities and logarithms. The third method can be solved with a calculator.

Because none of the three references give a thorough numerical example, a common one was generated for all three. Though the American Society for Testing Materials (ASTM) considers several experimental variations, only the more straightforward method of two minimum pressures is developed in depth. The ASTM result did not agree with the Morse and Swenson value. In this comparison, experimental data is not being used. Also, the ASTM approach used accommodates two minimum pressures, whereas the others consider only one.

The common example gave the following values for the REAL and IMAGINARY parts of the impedance, where the REAL part is related to the attenuation coefficient. This coefficient is not the surface absorption coefficient. Care must be given to note what specific physical quantity is being defined in the literature.

	REAL	IMAGINARY
Morse	1.2630	0.7499
Swenson	1.2630	0.7499
ASTM	1.8717	-0.1858

REAL = 0 (complete absorption into open space)
REAL = 1 (complete absorption at item)
REAL goes to infinity (complete reflection at item)

The speaker (acoustic source) and sample location, with general wavelength profile, are illustrated in figure 1. During data acquisition, the microphone is moved from x=XL down toward the speaker at x=0.

SPEAKER at X=0	X=XC	X=XB	X=XA	X=XL(face of sample)
PRESSURE:	MIN	MAX	MIN	

The system should follow these criteria in setup and operation:

1. Tube interior must be straight, smooth, with uniform cross-section and rigid enough to keep it from vibrating during testing.

2. Tube diameter D: For a given diameter, acoustic frequencies must be below the value f: $f < (0.586*C/D)$ C =sound speed = 1,128 fps at 20°C.

   In inches: $f < (7930./D)$

f (Hz)	D (in.)
100	79
500	16

   For a 4-in. diameter pipe, the highest frequency is 2,000 Hz, and 12,000 Hz for a 1-in. diameter. However, in practical use:

Inside diameter (in.)	Use frequencies (Hz) below
4	1500
1	6,000

3. Tube length L: For measuring at least two minimum pressures (if the ASTM method is used), and to avoid any measurement within one tube diameter from the speaker, the length should be: $L > [(0.75*C/f) + D]$

   For a frequency f=25 Hz, and D=4 in., L > 34 ft
   For a frequency f=50 Hz, and D=4 in., L > 17 ft
   For the other methods, a shorter tube would satisfy

4. Specimen: The specimen with its detachable holder must make an airtight fit at the tube end. The specimen should be flat and of the same cross-section of the tube. The holder should be a massive sound-reflective metal. Two or more specimens should be measured and the results averaged.

5. Speaker: The speaker may face directly into the tube or be mounted in an enclosure to the side if it is too large. Some absorbent material should be placed at the speaker end to avoid standing wave resonance. The speaker frequency should be determined to within 1%, or within 1 Hz.

6. Microphone: If the microphone is small enough, it can be placed inside the tube, and moved by a rod connection along the length of the tube. Or the microphone can be attached to a probe extended radially to the tube center from a longitudinally slotted tube. An axial position scale measurable to the nearest millimeter is needed.

A common example to determine material impedance is used for all three approaches.

METHOD A. Vibration and Sound (ref 1)

Experimental data needed: XL,XA,XB,PMIN,PMAX

This approach does not account for energy loss by sound waves through absorption in the tube.

X=0 = location of speaker
X=XL = distance from speaker to specimen face = 30 ft
X=XA =location of pressure MIN from specimen = 26 ft
X= XB = location of pressure MAX = 20.36 feet.
XLAM = wavelength = 22.56 ft (XLAM= C/f = 1,128 fps at 20C/ 50 Hz signal)
PMIN = pressure at XA = 0.1 Pascal or volt
PMAX = pressure at XB = 0.2 Pascal or volt
PI = 3.141592654

Find ALPHA and BETA from

BETA = 1. – 2.*(XL-XA)/XLAM = 1.-2.*(30-26)/22.56 = 0.64539 (1)

tanh (PI*ALPHA) = PMIN/PMAX = 0.1/0.2 = 0.5 (2)

This method is accurate when ALPHA falls between 0.02 and 0.5.

Before equation 2 is solved, the following definitions illustrate the relevance to the real impedance THETA that is used in ANSYS finite element attenuation:

ZETA = Average specific acoustic impedance of item at XL
ZETA[pressure/velocity] = ZM[force/velocity]/S (area of tube)
ZM = mechanical impedance of item at XL
XSI = Dimensionless acoustic impedance ratio at XL
XSI = Average specific acoustic impedance at XL /fluid characteristic impedance

XSI = ZETA/(R*C) = THETA – i*CHI = TANH(PI*ALPHA – i*PI*BETA) (3)
XSI = P/(U*R*C)

THETA = ACOUSTIC RESISTANCE (dimensionless) = real part of XSI.
CHI = ACOUSTIC REACTANCE (dimensionless)
R*C =fluid characteristic impedance (density * sound speed)
U = particle velocity

ANSYS employs the parameter MU:

MU = 1/THETA =R*C (of fluid)/Real part of material impedance = R*C/(R*C*THETA)

For MU = 0 = high impedance at item. Item boundary reflects

MU = 1 = all absorption at the item boundary
MU around E8 for open space

Solve equation 2 by a calculator. This gives PI*ALPHA = 0.549306
Solve equation 3 by complex exponentials in Morse FORTRAN program (table 1).

XSI = tanh S = sinh S/cosh S = (e**S-e**(-S))/(e**S + e**(-S))

where S = PI*ALPHA – i*PI*BETA

XSI is expressed in FORTRAN as complex natural exponents:

XSI = tanh S = [cexp(S) – cexp(-S)]/[(cexp(S) + cexp(-S)]

where XSI and S are declared complex.

For our case: XSI = tanh (0.5493 – i*2.0275), S = S(0.5493,-2.0275)

With this complex value for S in Morse FORTRAN program, the real part of XSI is THETA (the acoustic resistance) and the imaginary part of XSI is the negative of CHI (the acoustic reactance).

For this example: THETA = 1.2630, CHI= 0.7499

The absorption coefficient is easy to determine.

Find standing wave ratio: SWR = PMAX/PMIN

SWR = 1 = ALL ABSORBED (PLANE WAVE).
SWR = INFINITY = NOTHING ABSORBED.
R = REFLECTION COEFFICIENT = ((SWR-1)/(SWR+1))**2 = (1.3)**2= .111
IF R = 1, ALL REFLECTED.
AB = ABSORPTION COEFFICIENT = 1.- R = 0.888

METHOD B. Impedance from Principles of Modern Acoustics (ref 2)

Experimental data needed: XL,XA,PMIN,PMAX

1. Same physical setup as for METHOD A.
2. Find A from: A = (XA-XL)/XLAM where wavelength = XLAM = C/f, and C=1128 ft/s at 20°C, and atmospheric pressure.
3. Use f=50 Hz, XLAM= 22.56 ft, XL=30 ft, XA=26 ft.

4. A= (26-30)/22.56 = -0.1773 Minimum amplitudes relate to phase by: 4*PI*f*(XA-XL)/c = (2*n+1)*PI - PHA. (n=0,1,2,3...)

   PHA = phase angle of P(reflected)/P(incident).
   First min is related to phase PHA of wave:
   PHA = PI - 4*PI*F*(XA-XL)/XLAM = PI*(1-4*A)
   PHA = PI*(1 - 4(-0.1773)) = 5.3697 radians = 307.7 degrees
5. Find: (SWR-1)/(SWR+1) = 1/3 = Q (where SWR = PMAX/PMIN=.2/.1=2)
6. Complex impedance = Zo *[(1+Q*exp*(i*PHA)]/[1-Q*exp(i*PHA)]
   Complex impedance = Zo*[(1+Q*exp(i*5.3697)]/[1-Q*exp(i*5.3697)]
   Zo= R*C = 413 [kg/(m**2-sec)] (not needed)

Use complex exponential:

THETA - i*CHI = Complex impedance/Zo = [1+Q*expS]/[1=Q*expS], where complex S only has an imaginary part =(0,5.3697)

Then XSI = THETA - i*CHI = [1+Q*CEXP(S)]/[1-Q*CEXP(S)].

Use the Swenson FORTRAN program to determine THETA.

Result: THETA = 1.2630 CHI = 0.7499

METHOD C. ASTM Method C384-90A (one maximum and two minima)

Data needed: XLAM,XA,XC,PMIN1,PMAX,PMIN2, with optional calibration. Same convention of x=0 at speaker location is used as in METHODS A and B.

This method requires at least two minimum pressures be detected.

Here the tube LENGTH should be: L > 0.75*XLAM + D. (D=tube diameter)

Procedure: (Use XLAM(wavelength) = 22.56 ft for a 50 Hz signal).

1. Place METAL END REFLECTOR at x=XL, the position where the specimen face will be.
2. Measure XAC, the x coordinate for first minimum.
3. Measure XCC, the x coordinate for second minimum.
4. XL calculated as = (3*XAC- XCC)/2.
5. Place front of SPECIMEN where front of the reflector had been.
6. Determine coordinates, XA and XC, of the first two minimum pressures from specimen.
7. Calculate: D1= XL-XA = distance of first minimum to face of specimen.
8. Calculate: D2 = XA-XC (which is about half wavelength).
9. Adjust signal so minimum pressures are more than 10 dB above the noise.
10. Don't measure MAX at face of specimen.
11. PMIN1 = first min pressure. PMAX = first maximum pressure. PMIN2 = second minimum pressure. These can be in Pascals or volts (one or the other).

12. Transform these pressures to dB:
13. PMIN1 =20.*log(PMIN1/29E-6). PMAX=20*LOG(PMAX/29E-6)
14. PMIN2 =20.*log(PMIN2/29E-6).
15. Calculate: L1= PMAX-PMIN1
16. L2= PMAX-PMIN2.
17. L0= L1 + (L1-L2)/2
18. Calculate Ko= 10**(L0/20)
19. Calculate: M=(Ko + 1/Ko)/2
20. Calculate: N = (Ko - 1/Ko)/2
21. Reflection phase angle: PHI (radians) = (D1/D2 - 0.5)

Definitions for r (real part of impedance), and x (imaginary part of impedance):

1. The impedance ratio: z/(rho*c) = r/(rho*c) + i*x/(rho*c)
2. Where: THETA = 1/(M-N*cos PHI) = r/(rho*c) = resistance ratio
3. Where: CHI = (N* sin PHI)/(M-N*cosPHI) = x/(rho*c) = reactance ratio

Notes:

1. If D1/D2 < 0.5, phi is negative, and x/(rho*c) is negative.
2. Specific admittance = 1/z = y
3. Specific normal acoustic admittance ratio = y*rho*c
4. Admittance ratio (y*rho*c) is larger than specific admittance (y) by about 400 in SI units.

Example: No specimen is present, only the metal reflector.

XA=30-XLAM/4= 24.36
XC=XA-XLAM/2 = 13.08
XL=(3*XA=XC)/2 = 30 ft

The specimen is placed in position:

XA=26
XC=14.72
D1=XL-XA=30-26=4 ft
D2= XA-XC =11.28 ft

Suppose pressures are: PMAX, PMIN1, PMIN2 = 0.2, 0.1, 0.09. Convert to dB gives: 76.77, 70.75, 69.83 = (20*log(P(peak)/29E-6)

L1 =PMAX - PMIN1 = 76.77 - 70.75 = 6.02 dB
L2 =PMAX - PMIN2 = 76.77 - 69.83 = 6.94 dB
L0 = L1+(L1-L2)/2 = 5.56 dB
K0 = 10**(L0/20) = 1.897
M = (K0+1/K0)/2 = 1.2122
N = (K0-1/K0)/2 = 0.6852
PHI = (D1/D2 -0.5) = -0.1454 RADIANS
THETA = 1/(M-N*cos PHI) = 1.8717
CHI = (N*sin PHI)/(M-N*cos PHI) = -0.1858

DETERMINATION OF SINGLE PULSE SPECTRAL AND ENERGY CONTENT

The following procedure covers the essential steps in pulse analysis, with illustrations made with Vu-Point 3 software for an experimental impulse noise.

1. Scale the signal to actual Pascal pressure, and plot it linearly in pressure and time. Determine the peak pressure.

2. Determine the RMS value of the signal. This is generally not well defined for pulse signals. An arbitrary interval is from signal start to the time where the signal finally stays within the boundary of plus and minus ten percent of the peak pressure (20 dB down from the peak).

3. In preparation for the FFT (fast Fourier transform), use a Hanning (A=0.50) or a Hamming (A=0.54) filter on the N data points: [A+(1-A)*cos(2*pi*n/N)]. Also remove linear trends in the signal. For a pulse with ample time before and after the pulse, use or non-use of the filter is indifferent. This ample time should be present to avoid signal distortion if the filter is used. Obtain the FFT of the signal.

4. Obtain the real spectral ENERGY DENSITY, FF*, which is the FFT of the auto-correlation function.

5. Integrate the FF* to obtain a step function showing relative energy content as a function of frequency.

6. If there are dominant frequencies, find the percent contribution to intensity (energy) from each dominant frequency by dividing each step increase by the overall maximum. First subtract, if present, any energy level around zero frequency.

This procedure is illustrated by these figures:

Figure 2 is a time signal from 440 ms to 820 ms. The sampling rate (DT) is 20.833 μs (48,000 data/s). The recording time (T) is 0.38 sec. Number of data points (N) is = T/DT = 0.38/ 20.833 μs = 18,240.

If all the data points are used in the FFT, then in the FFT:

Frequency resolution (DF) = 1/T = 1/(N*DT) = 2.63 Hz
Frequency range (F) = 1/(2*DT) = 24,000 Hz

If the analysis program uses a number of data points N =2**n (n=..12,13,14..), then the program chooses n to have 2**n close to N. As examples:

2**14 = 16,384
2**15 = 32,768
2**16 = 65,536
2**17 = 131,072

The program may choose 2**14 points and eliminate data from the 18,240, or use 32,768 and pad the program with data. The frequency resolution (DF) of the resulting FFT can be checked.

Figure 3 is the FFT of figure 2. The frequency resolution is 1.46 Hz. Here, the program internally chose 2**15 data by adding 14,528 points. A short recording time T for a pulse should be avoided. Dimensions are [Pascal/Hz].

Figure 4 is the real spectral energy density of the signal as a function of frequency. Dimensions are [Pascal/Hz]**2.

Figure 5 is the integral of the spectral density, showing accumulative energy, as a function of frequency. Dimensions are [Pascal]**2.

Figure 6 is the INVERSE of the FFT in figure 2. This fairly well reproduced the time signal. Dimensions are [Pascals].

Figure 7 is an FFT with the Hanning filter of the immediate pulse itself. Though it resembles the FFT in figure 3, its frequency resolution is coarse at 5.06 Hz. The recording time T of only the pulse length is insufficient.

Figure 8 is the INVERSE of the FFT in figure 7. Use of the filter has appreciably distorted the pulse representation.

Figure 9 is the original time pulse of figure 2, but with a broader time base of 2 sec. The number of data N here = 2/20.833 μs = 96,000.

Figure 10 is the FFT with a Hanning filter of figure 9. The frequency resolution is 0.366 Hz. The program chose 2**17 and added 35,072 points to the available 96,000 for a total of 131,072.

Figure 11 is the FFT of figure 9, but no Hanning filter is used. This result is essentially the same as figure 10, since the signal in figure 9 is isolated in time in both directions.

ACOUSTIC SIGNAL CONVERSIONS

1. Determination of peak pressure of discrete signals in a composite signal:

   If the fast Fourier transform is not cluttered and shows a signal composed of discrete frequencies, then the peak pressure in Pascals of each discrete frequency is obtained by multiplying by two (waves are positive and negative by FFT algorithms) the integrated area over each frequency occurrence in the FFT. This procedure is NOT used if a cluster of frequencies occurs around the various dominant frequencies.

2. Determination of pressure at distances from the source:

Estimating acoustic signal dB at various distances from the source must be done carefully. The effect of humidity, fog, ground hardness or porosity (snow), ground reflection and temperature gradient in the region above the ground, and size of the emitting source, have been reported, but with limited usefulness for some engineering applications. The final recourse is to do calibrated acoustic testing with sources and along distances where subsequent investigation is relevant.

The calculation of energy decreasing as inverse distance squared (pressure decreasing as inverse distance) from a point source, into a non-attenuating sphere, is simple, but not realistic:

This assumes intensity varies as inverse distance squared along a radial line between source and microphone:

I2 = I1 * (R1/R2)**2
I1 = intensity at radial distance R1 (microphone location) from source.
I2 = estimated intensity at radial distance R2 from sound source.
If I1 = 1.79 W/m**2 or 122.54 dB at R1 = 25 meters, then at R2=100 meters:
I2 = 1.79 (25/100)**2
I2 = 0.11188 W/m**2.
dB(at distance R2) = 10 log (.11188/E-12) = 110.49 dB

However, if the pressure at the microphone is calibrated at two or more places, the distance-pressure data can be fit to a geometrical regression to obtain coefficients A and B to predict pressure at a location X that is in a reasonable vicinity where calibration was done: P=A*(X**B))

If two calibration points are made: P1= 2 Pascals at x1= 3ft, and P2= 0.3 Pascal at x2=10 ft, then the coefficients are:

B= log(P2/P1) / log(x2/x1) = log(.3/2)/log(10/3) = - 1.576
A= P1/(x1**B) = 2 /(3**-1.576) = 11.297

At a distance X=15 ft, the estimated pressure is P = 11.297*(15**-1.576) = 0.158 Pa. For three or more points, a standard curve fit is used to generate A and B for the pressure fit.

3. Adding dB from composite sine-type signals:

Four frequencies, each with a peak Pascal pressure amplitude, are added to form a composite signal:

Frequency	Input amplitude Pascal	dB
1000	10	110.75
1500	5	104.73
2000	1	90.75
4000	5	104.73

dB is from: dB= 20 log (P/29E-6) [P in peak Pascals]

This composite signal has a Pascal RMS of 8.69. The signal dB= 20 log(P[RMS]/20E-6) = 20 log(8.69/20E-6) = 112.76 dB

If the peak amplitudes (or dBs of the COMPOSITE signal) are known, take the highest value, and find the difference of the others:

Highest dB = 110.75
2nd-highest: 110.75-104.73 = 6.02 =DIF DB
3th-highest: 110.75 - 104.73 = 6.02 =DIF DB
4rd-highest: 110.75 - 90.75 = 20.0 =DIF DB

Use the quadratic expression, ADD DB, to find what additional dB is added to the dB of the dominant frequency, by each frequency other than the dominant one:

ADD DB = 3.0 - 0.427*(DIF DB) + 0.0153* (DIF DB)**2

In this case:

2nd: ADD DB = 3.0 - 0.427(6.02) + 0.0153*(6.02)**2 = 1.0
3rd: ADD DB = 3.0 - 0.427(6.02) + 0.0153*(6.02)**2 = 1.0
4th: ADD DB = 0. If DIF DB is 14 or more, there is no contribution.

Total dB of signal: 110.75+1+1 = 112.75. This is close to the known dB of the entire signal.

4. Pressure-Power-dB Conversions:

Pressure (volts) (multiplying increment)	Power (watts) (multiplying increment)	dB (additive increment)
1.0	1.0	0.0
1.122	1.26	1.0
1.26	1.59	2.0
1.414	2.0	3.0
2.0	4.0	6.0

Pressure (volts) (multiplying increment)	Power (watts) (multiplying increment)	dB (additive increment)
2.828	8.0	9.0
3.162	10.0	10.0
4.0	16.0	12.0
5.0	25.0	14.0
10.0	100.0	20.0
100.0	10,000.0	40.0
1,000.0	1,000,000.0	60.0

A 12 dB increase of an 80 W signal is 16*80 = 1280 W.
A 12 dB increase and 3 dB decrease is 8*80 = 640 W.
A twofold increase in voltage results in a 6 dB signal increase.
Conversely, the increments can be dividing and subtractive.

5. Intensity, Pressure (Peak, RMS) relations:

Effect	Intensity (dB)	Intensity (W/m**2)	Peak pressure (Pascal)	Peak pressure/SQRT2 (Pascal RMS)
--	194	25.1E6	143,340.0	101,360. (atm,14.7psi)
--	191	12.6E6	101,360.0	71,700.0
--	180	1E6	28,900.0	20,400.0
--	160	10,000.0	2890.0	2040.0
--	140	100.0	289.0	204.0
Pain	120	1.0	28.9	20.4
Riveter	95	0.0032	1.631	1.15
Train	90	0.001	0.91	0.644
Busy Street	70	E-5	0.091	0.0644
Quiet radio	40	E-8	0.0029	0.002
Leaves	10	E-11	9.1 E-6	6.4 E-6
Threshold	0	E-12	28.9 E-6	20.4 E-6

6. Acoustic wavelengths for 20C and 1 atmosphere:

Wavelength for frequencies based on speed=344 m/s = 1128 ft/s:

f (Hz)	Wavelength
20	56.4 ft
100	11.3 ft
500	27.1 in.
1000	13.5 in.
5000	2.7 in.

7. Direct conversions:

dB = 10 log (I/E-12) [I in W/m**2]
dB = 20 log (P/20E-6) [P in Pascals, RMS]
dB = 20 log (P/29E-6) [P in Pascals, peak]
P(Pascals RMS) = 20 E-6 * antilog(dB/20) = 20 E-6 *(10**(dB/20))
P(Pascals peak) = 28.9E-6* antilog(dB/20)
I[W/m**2] = ((Pascals,RMS)**2)/400
I[W/m**2] = (Pascals,peak)**2/(2*D*V) = (Pascals,peak)**2/825.6 (air)
1 Pascal = Newton/m**2 = 10 dyne/cm**2
1 psi = 6895 Pascals
1 joule = E7 ergs = E7 dyne-cm =Newton-meter
1 atm = 1.01325E5 Pascals = 14.7 psi=1.013E6 microbars (dyne/cm^2)
1 bar = E5 Pascals
1 Watt = joule/sec
1 Newton = E5 dynes

8. Technical notes:

Intensity (RMS) =[Pressure(RMS)]**2/(D*V)
D= air density (kg/m**3)
V= sound velocity in air (m/s)
D*V = impedance (at 20°C and 1 atmosphere):
D*V (for air) = 1.2 [kg/m**3] * 344 [m/sec] = 412.8 [kg/(m**2-sec)]
D*V (for water) = 1000 * 1480 = 1,480,000 [kg/(m**2-sec)]

For the same acoustic pressure, the intensity in air is (1.48E6/413) = 3,580 greater than in water.

For the same frequency and particle displacement, the intensity in water is 3,580 times greater than in air.

The displacement amplitude of air = A =P(max)*V/(2*PI*B*F).
P(max) = maximum (not RMS) pressure amplitude. Use 30 Pascals.
B = air adiabatic bulk modulus at normal conditions = Gamma*air pressure = 1.4 * 1.013E5 Pascals = (142,000 Pascals).
PI = 3.14159 and use F=1000 Hz.
Gives: A = 1.2 E-5 meters = 0.012 mm (extremely small).

Wavelet transforms is an alternate method to the FFT for obtaining good frequency resolution (long recording time) up to higher frequencies (short sampling time) for pulses and portions thereof. Here, a signal region is expanded on an 'analyzing wavelet' function that is relevant to the signal under consideration. Examples of these transforms are not included in this report.

AURAL LIMITS BY MIL-STD-1474C

A distinction is made between steady state and impulse noise. If a steady state noise is below 85 dB or impulse noise is below 140 dB (determined from the peak pressure), no protection is needed.

Impulse noise is characterized by two parameters: the peak pressure level in dB (dB = 20*log (Peak Pascal)/ 29E-6) and the effective time duration (B-duration) of the sound in milliseconds. The dB and B duration lie in some area of figure 12 and by table 4, three of these areas: X, Y, and Z require specific hearing protection.

Table 4
Maximum daily exposure to impulse noise

Region	No protection	Either plugs or muffs	Both plugs and muffs
W	----------------------------	Unlimited Exposure	----------------------------
X	0	2,000	40,000
Y	0	100	2,000
Z	0	5	100

A single exposure consists of either a single impulse for non-repetitive system, or a burst as for an automatic weapon.

The sequence of steps to determine and A and B durations will be illustrated with the impulse time trace of figure 13, which is the same as figure 2. The 20 dB intervals are marked.

1. Find the peak pressure (voltage) of the time signal. Peak = 0.215 V.
2. Find A, time for pressure to go from zero, rise to first principal positive peak, and return to zero. A= 546 ms – 536 ms = 10 ms.
3. Set two bands, L+ and L-, at 20 dB below the peak (10% of peak), and its mirror image about zero level. L+ =0.0215, L- = -0.0215 V.
4. Find B, time from first zero, to the time when pressure finally stays between L+ and L-. B = 633 ms – 536 ms = 97 ms. Q = extreme time point 633 ms.
5. Find T1. First estimate is T1=5*A = 5*10 = 50 ms.
6. However, T1 must be equal or less than 30 ms. And also, if T1 > B, then start reducing the integer from 5 to 4,3,2...
7. Sequence: T1=4*A = 40 ms (Still greater than 30 ms).
8. Sequence: T1=3*A = 30 ms (At least equal to 30 ms. Stop.)
9. Further test: Find 0.3*B. 0.3*B = 0.3*97 = 29 ms.
10. Now if T1 is equal or greater than 0.3B, then B = duration of primary portion.
11. But if T1 is less than 0.3B, find the first point past 0.3B during which pressure lies entirely between L+ and L-. Let this time be Q. B duration is then Q-536ms. In this case, T1 > 0.3B.
12. If there are any subsequent fluctuations that vary outside the limits of L+ and L-, and whose time duration is greater than 10% of B, then add this time duration onto B. For this trace, there are none and the B duration is then 97 ms.

The transformation of peak volt to Pascals was 0.215*12580.=2700 Pascal(peak). Then dB=20*log(2700/28.9E-6) = 159.4 dB. By figure 12, this dB and B duration fall into region X.

BEAT FREQUENCY FROM MULTIPLE ACOUSTIC SOURCES

The usual approach of considering only two speakers is expanded into a more general arrangement of many acoustic sources on a panel perpendicular to the ground, and having microphones located anywhere in a region in front of this panel array. Some calibration must first be made for each speaker at various distances to obtain at least a geometric regression fit for sound received at each mike. The input data needed, and results, are calculated by the beat frequency FORTRAN program in table 5.

The acoustic sources are considered located on an X,Z plane, perpendicular to the ground, and the microphones are located anywhere at XM,YM, and ZM, where YM is the mike ground distance from the sources.

The program assumes there is complete mixing of the frequencies, though the degree of mixing is determined experimentally. The frequency content at the location of a microphone in the field is calculated by the program. This content is independent of the location, though there is a phase difference between various physical locations.

The following definitions are needed to describe the system and input to the program:

Xi= x coordinate of the ith speaker in X,M plane;
  i=1,2,3,4...up to as many speakers as are present.
Zi= z coordinate of the ith speaker
fi = frequency of the ith speaker (frequency can be any explicit function of time)
XMj= x coordinate (lateral) of the jth mike in the vertical XM,ZM plane
YMj= y coordinate (axial) of the jth mike from the plane of speakers.
ZMj= z coordinate (vertical) of the jth mike in the vertical XM,ZM plane

C = speed of sound = 1128 fps at 20 C.
Rij = actual straight line distance from ith speaker to the jth mike.
t = time
PI=3.141592654
i= square root of minus one

The acoustic signal that <u>arrives at the mike</u> has a peak pressure Si, and is described by:

Signal at jth mike from the ith speaker = Si *exp[i*2*PI*fi*(t+Rij/C)]

For absolute pressure, the mike must be calibrated with the speakers, operating at some constant frequency. For a generic expression for pressure as a function of distance, each speaker is calibrated at two or more locations. The programs uses the regression coefficients and calculates the signal strength, Si, at the jth mike from the ith speaker: Si = A * (Rij ** B) where A and B are obtain from the curve fit. If only two calibration points are obtained, as 15 Pascals at a distance of 5 ft, and 2 Pascals at 30 ft, then B = ln(2/15) / ln(30/5) = -2.01/(1.79) = -1.14, and A = 15/(5**B) = 94.

The frequency content is added to the signal strength, and the signal is now expressed in cosines and sines from the complex form:

Signal = Si[cos(2*PI*fi*(t+Rij/C)) + i * sin(2*PI*fi*(t+Rij/C))]

These are essentially vectors, and can be added as vectors. Figure 14 shows the addition of three such sources and the method is expanded in table 5.

The amplitude of the resultant vector is Ssum, and the phase angle is the inverse tangent of the ratio of the imaginary and real components. The straight-line distance between any speaker j and the mike i is:

Rij = SQRT(Q*Q + (DELX*DELX))

where Q = SQRT(YMij*YMij + (DELZ*DELZ))
DELX = XMij - Xi
DELZ = ZMij - Zi

Table 5 is the FORTRAN listing of the program illustrating three sources:

f1=2000 Hz
f2= 2100 Hz. If this frequency were amplitude modulated at 30 Hz, it would be expressed as f2= 2100* sin(PI*60*t), but such an expression must be recompiled in the program.
f3= 2150 Hz

For these three acoustic sources, and for <u>complete</u> signal mixing, the following figures document the results:

Figure 15 is the composite time signal at mike 1 for two source signals: 2000 and 2100 Hz.

Figure 16 is the FFT of figure 15, which extracts the beat of 100 Hz and harmonics.

Figure 17 is the composite time signal at mike 1 for three source signals: 2000, 2100, and 2150.

Figure 18 is the FFT of figure 17, which extracts the beat of 50, 100, and 150 Hz and harmonics.

An experimental beat frequency from 2000 and 2100 Hz sources is given in figure 19. The total recording time is T=2 sec, and sampling time is DT=20.833 μs. The frequency range is 1/(2*DT) = 24KHz. Instead of choosing the 96000 data points (T/DT), for a frequency resolution of 0.5 Hz (DF=1/T), the VuPoint3 program chose 2**17 data points (131072), for a frequency resolution DF=0.366 Hz.

The FFT of the positive signal is shown in figure 20. Whether a Hanning or no filter is used is irrelevant, except for magnitude. The sum and difference frequencies are extracted. Figure 21 is a finer detail of the carrier frequencies in the FFT. Figure 22 is the integral of the auto-correlation function, graphing the cumulative energy of the various frequencies. The zero frequency level is approximately 32% of the final sum in figure 22. When this zero level is subtracted out of the cumulative sum, the percent energy contained in the various frequencies becomes:

Frequency (Hz)	Energy (%)	
100	11	
200	1	
2000	37	
2100	38	
4025	1	
4100	11	
4210	1	Total=100%

MODAL FREQUENCIES IN AN ENCLOSURE

The standard expression for the discrete frequencies, and the number of each, has been reduced to a FORTRAN program in table 6 for a rectangular closed structure. For a closed reflecting room, the fixed wavelengths that can exist are basically given as

W=2/[SQRT((Nx/Lx)**2 + (Ny/Ly)**2 + (Nz/Lz)**2)]

where the program varies the integers Nx,Ny,Nz independently from 0 to 38. Lx,Ly,Lz are the room dimensions. The longest wavelengths go as twice the room dimensions, and for one open end they go to four times the dimensions. Figure 23 plots the specific frequencies that can exist in a 6.6 x 9.3 x 20.6-ft room, and the number of each up to a frequency of 220 Hz.

To illustrate the capability of finite element determination of acoustic properties in an enclosure, a room with the dimensions described in figure 23 is driven by an acoustic source in one corner of the room at 75 Hz with a peak to peak signal of two Pascals. Figure 24 plots the pressure contour on the viewed walls at one instant in time. The resulting pattern is not simple.

The dB pattern of this pressure is plotted in figure 25. To illustrate the pressure along a diagonal extending from the source to the other end of the room, figure 26 shows three locations where pressure nulls exist. This illustration is for a closed room with reflecting walls. Variations in closure and wall absorption are readily treated.

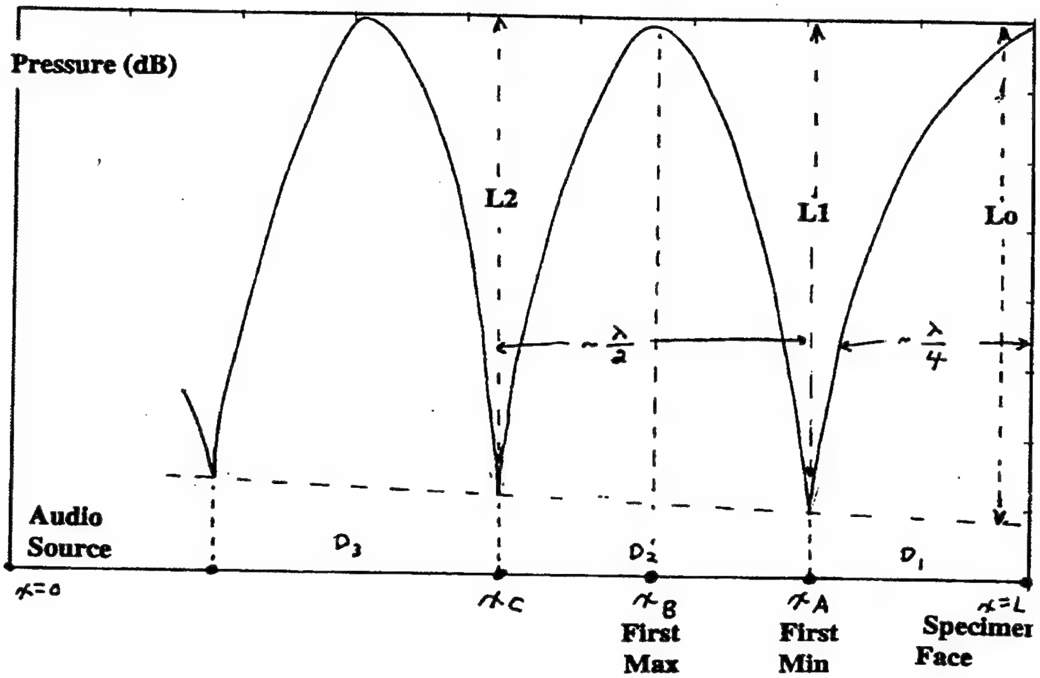

Figure 1
Pressure profile in acoustic impedance tube

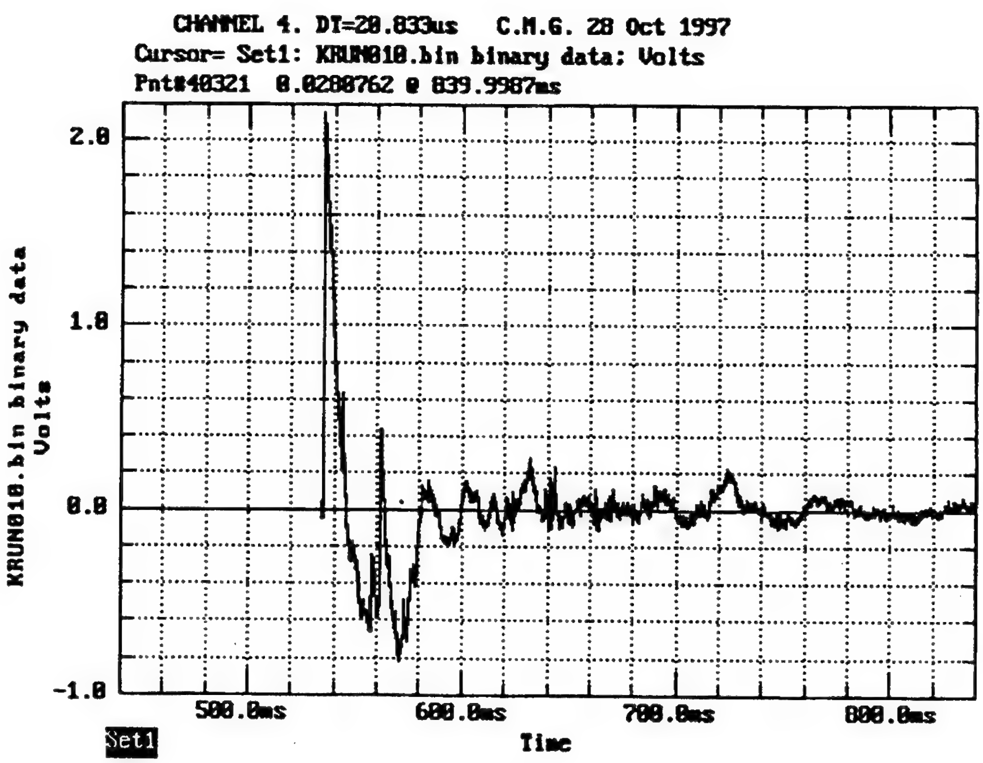

Figure 2
Acoustic signal in time domain

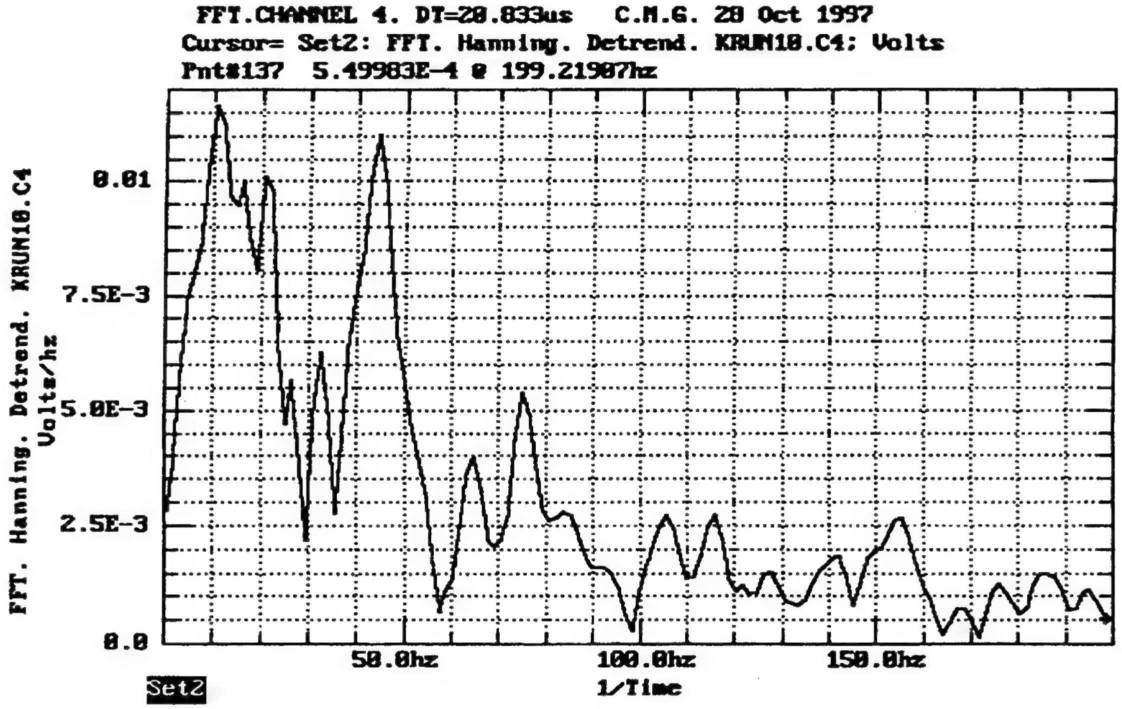

Figure 3
Fast Fourier transform of time signal pulse

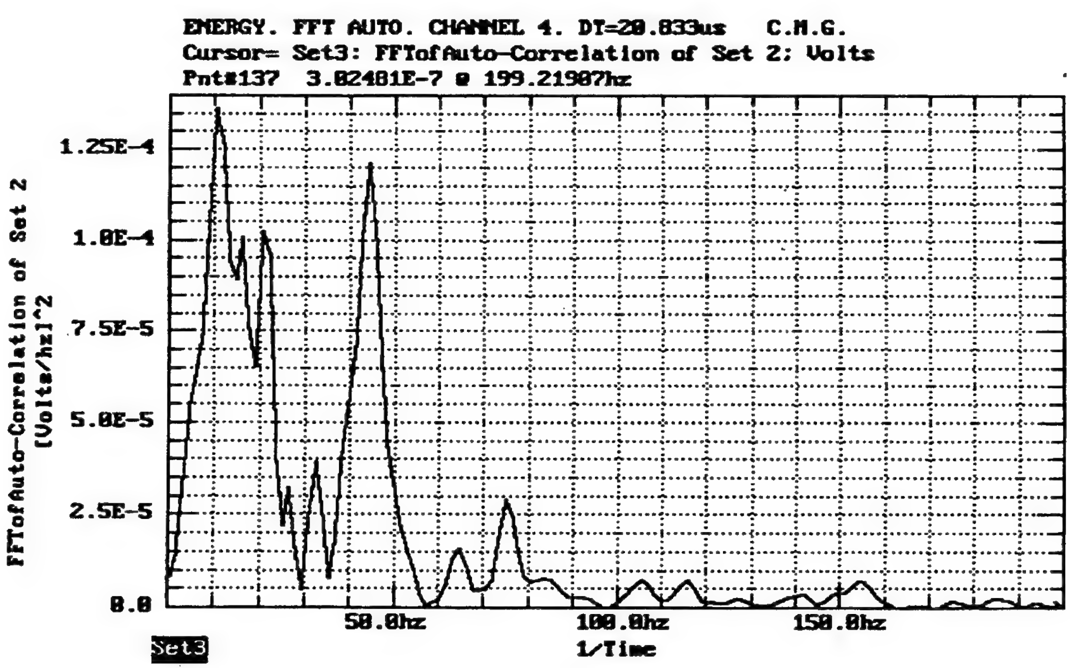

Figure 4
Energy spectral content of signal

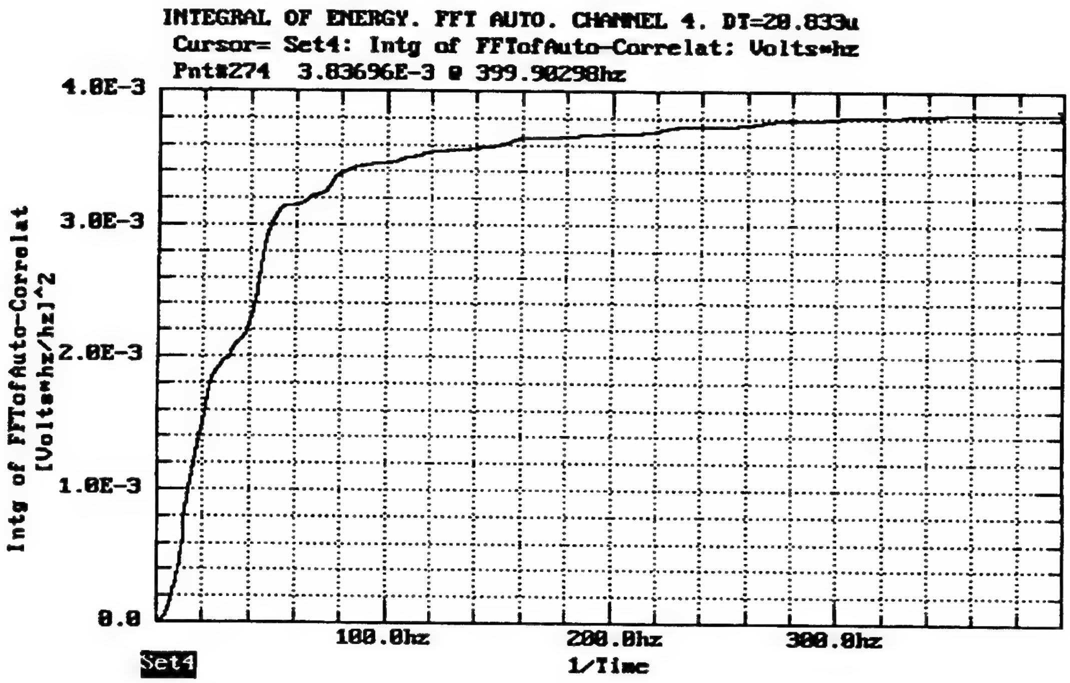

Figure 5
Integral of energy content of signal

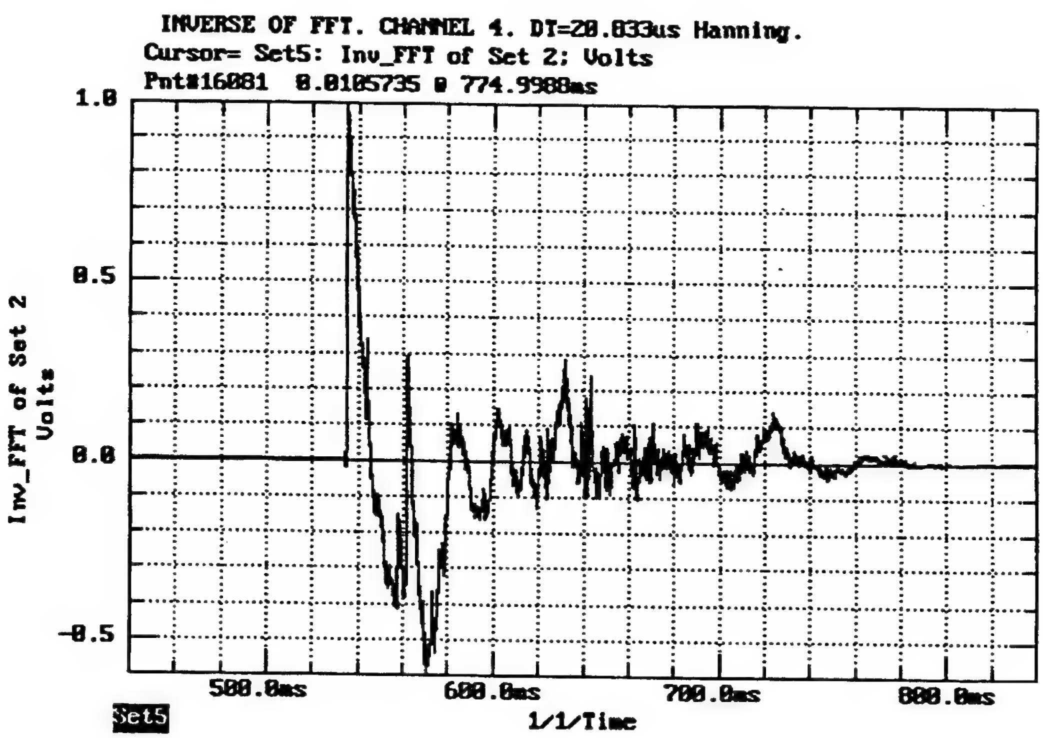

Figure 6
Recovered time signal by inverse FFT

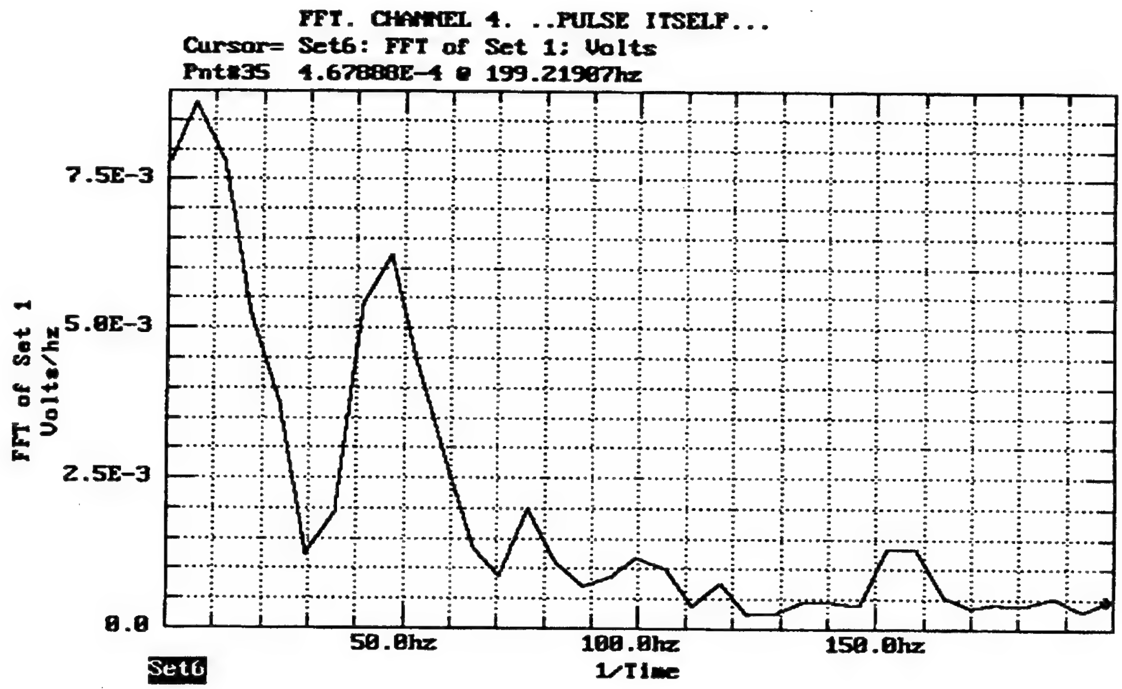

Figure 7
FFT of time signal pulse without time extension

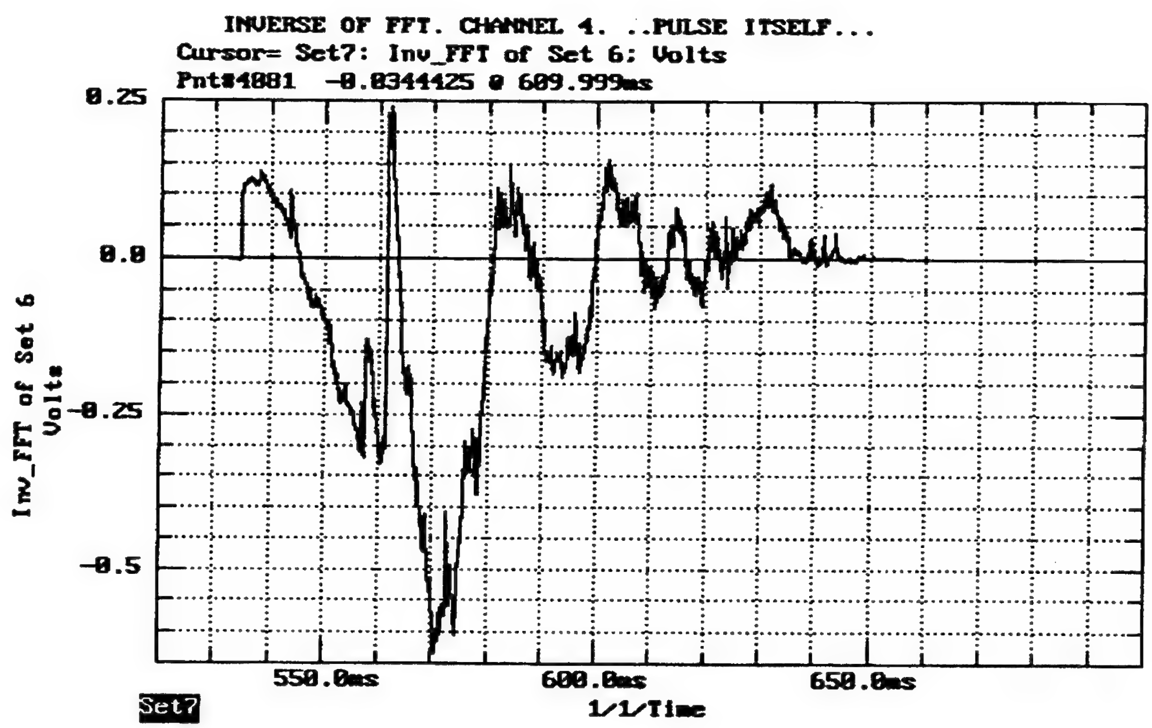

Figure 8
Inverse FFT of pulse signal without time extension

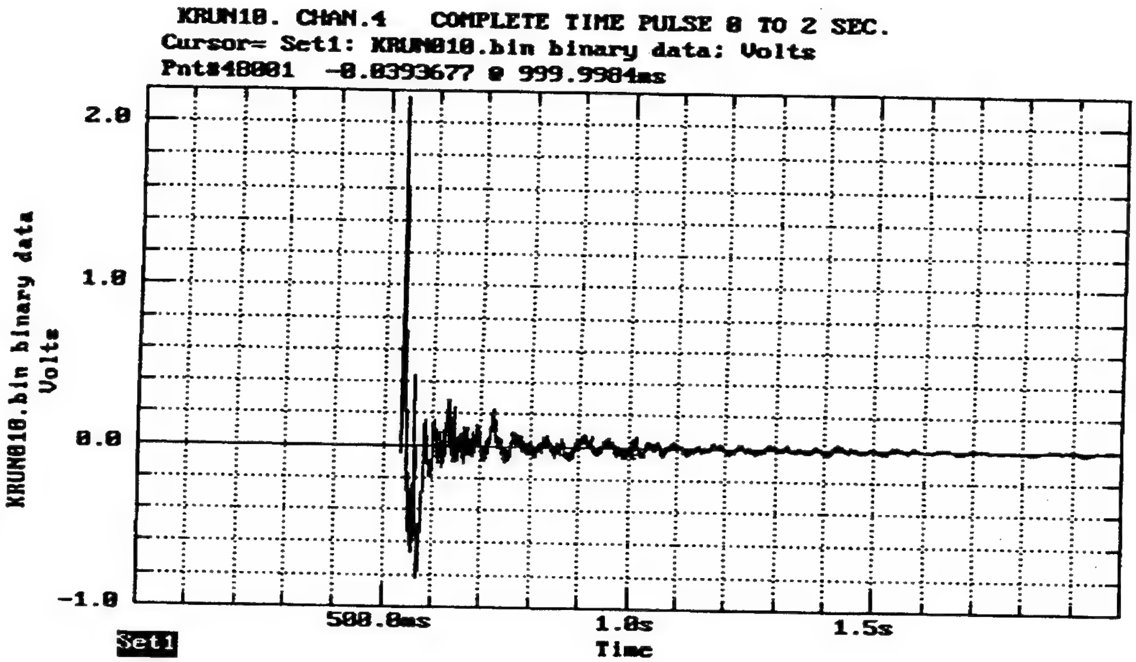

Figure 9
Acoustic pulse with long time duration

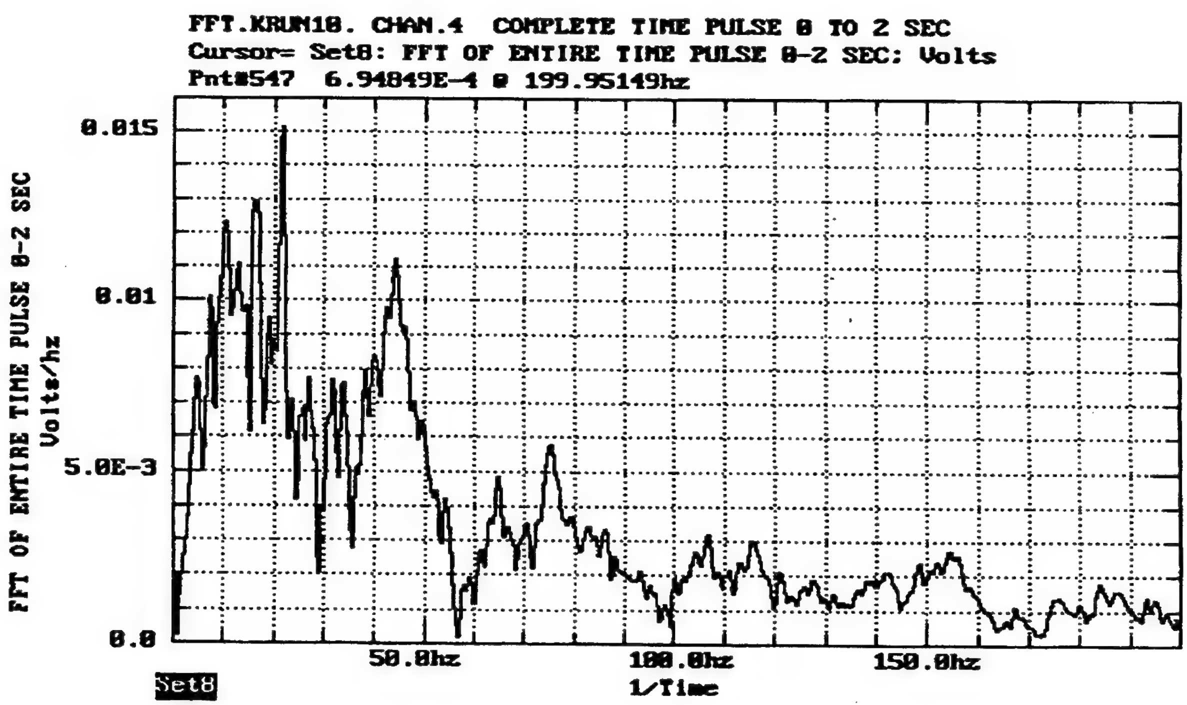

Figure 10
FFT of pulse with long time duration

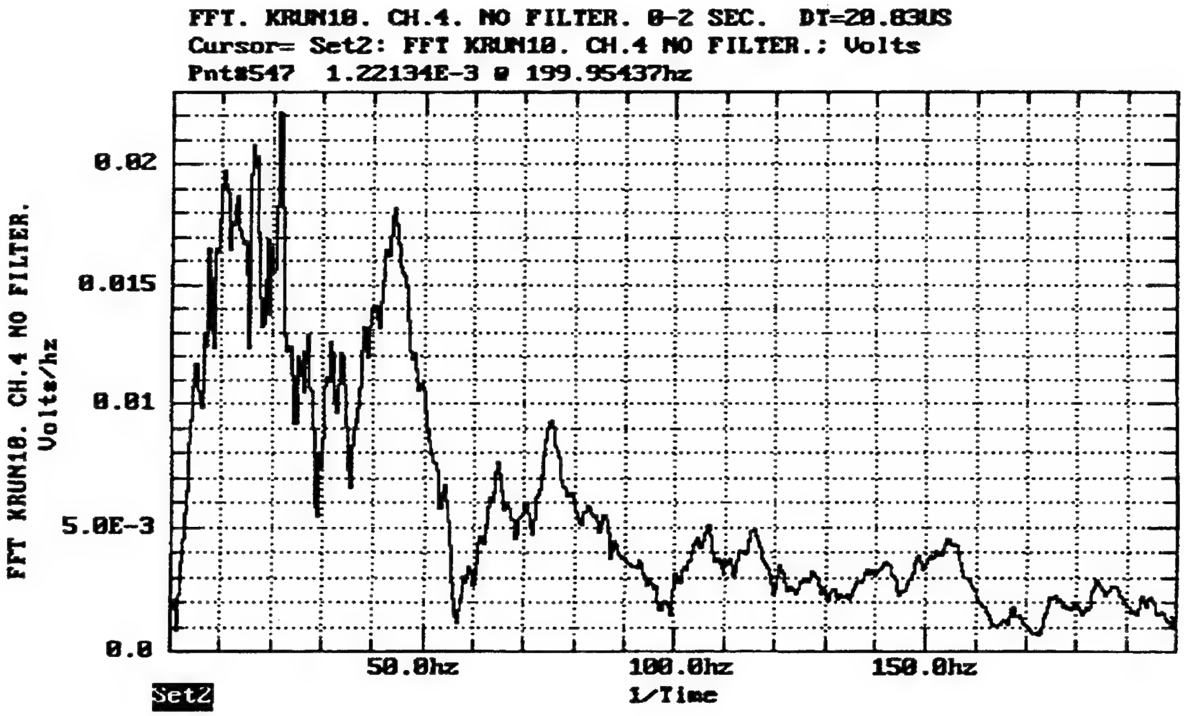

Figure 11
FFT of pulse with long time duration (no filter used)

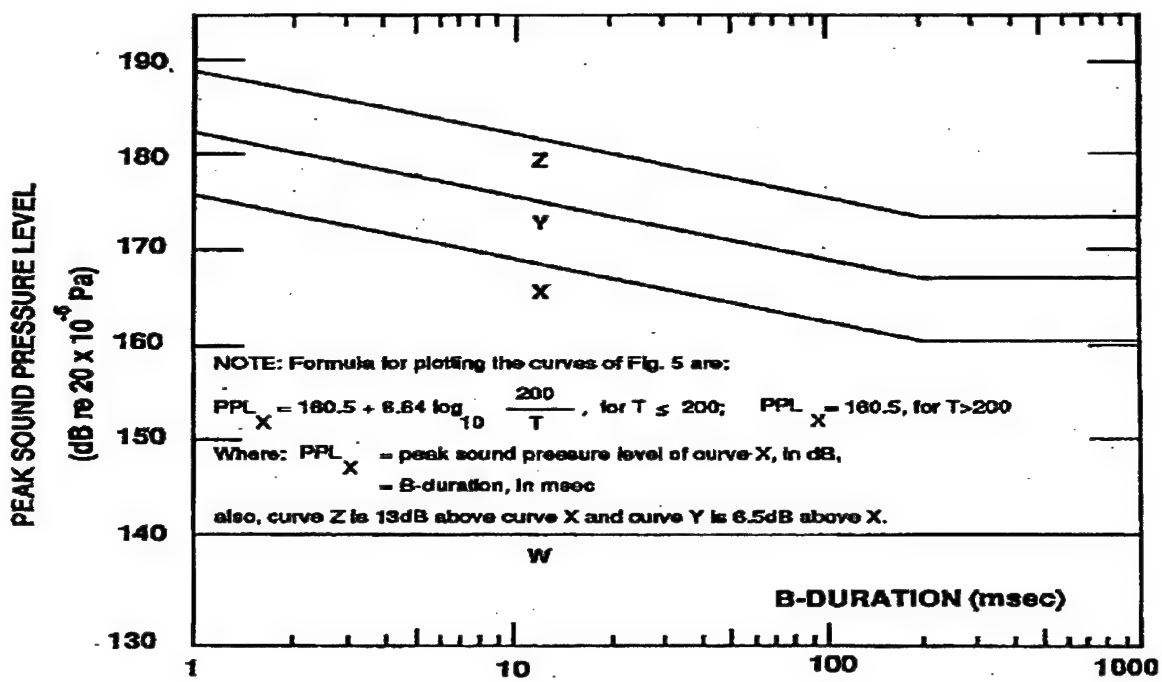

Figure 12
Regions of peak sound level and B duration for impulse noise

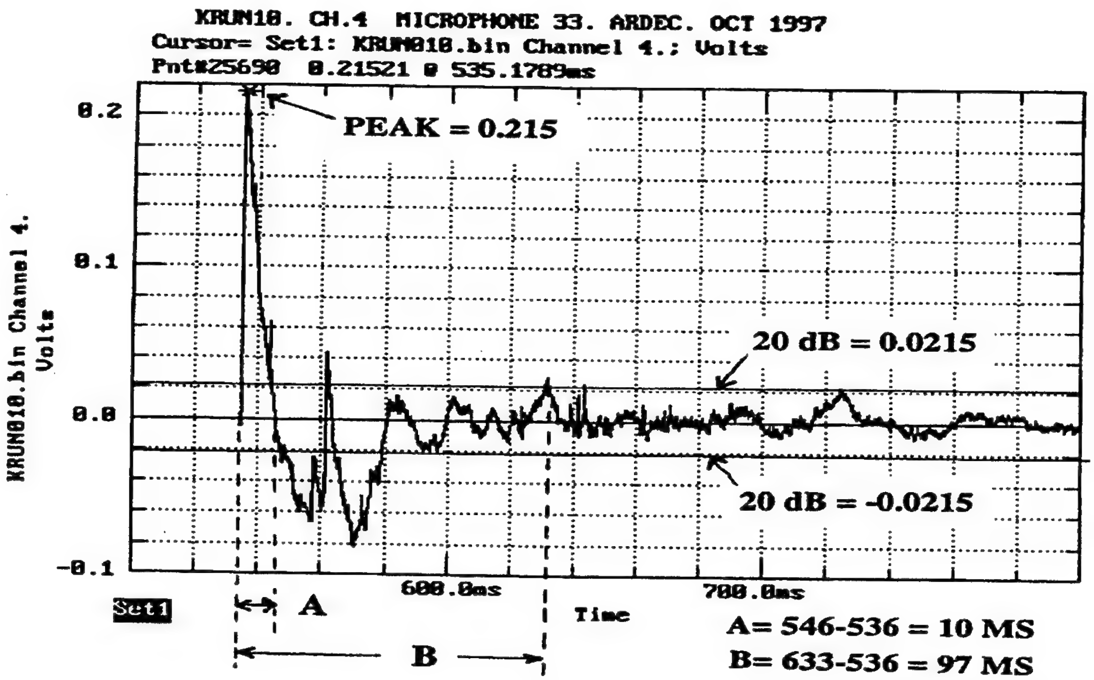

Figure 13
Acoustic pulse for B duration determination

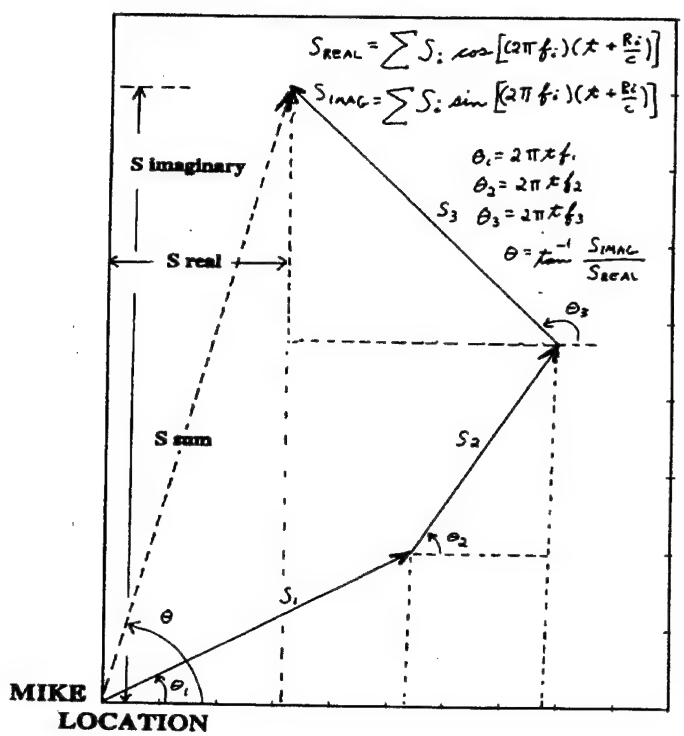

Figure 14
Frequency mixing at microphone for beats

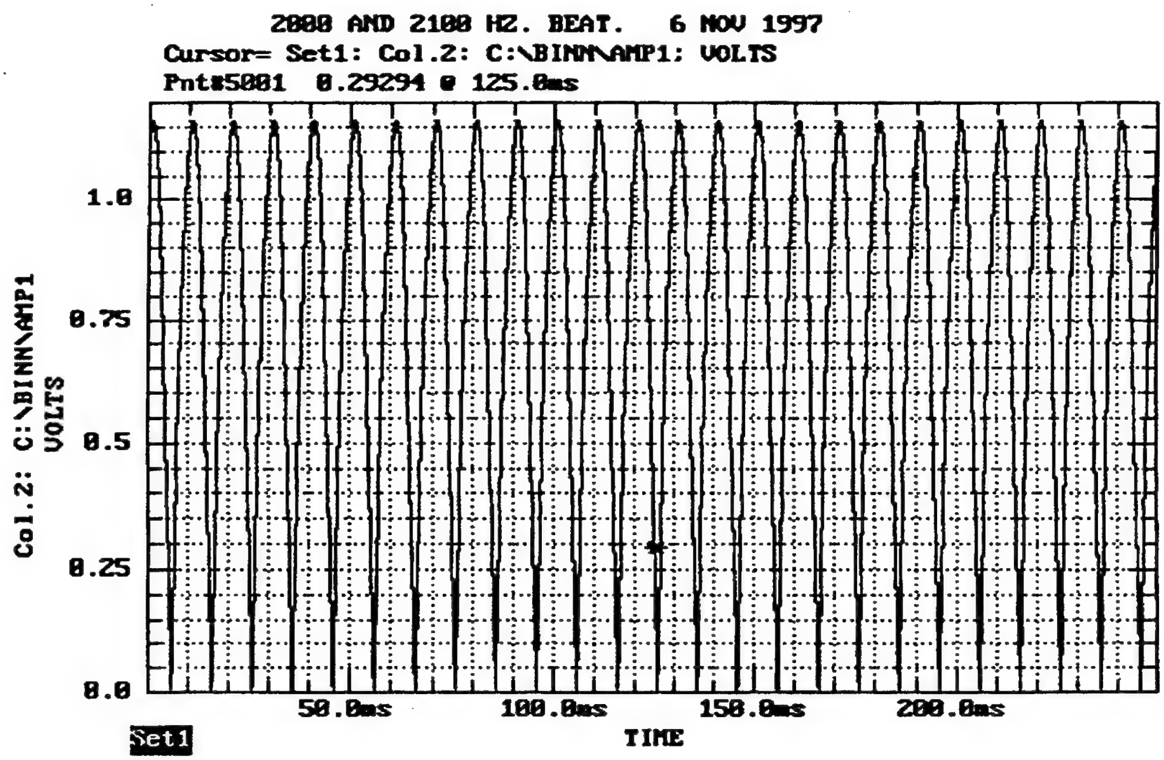

Figure 15
Composite time signal for 2000 and 2100 Hz

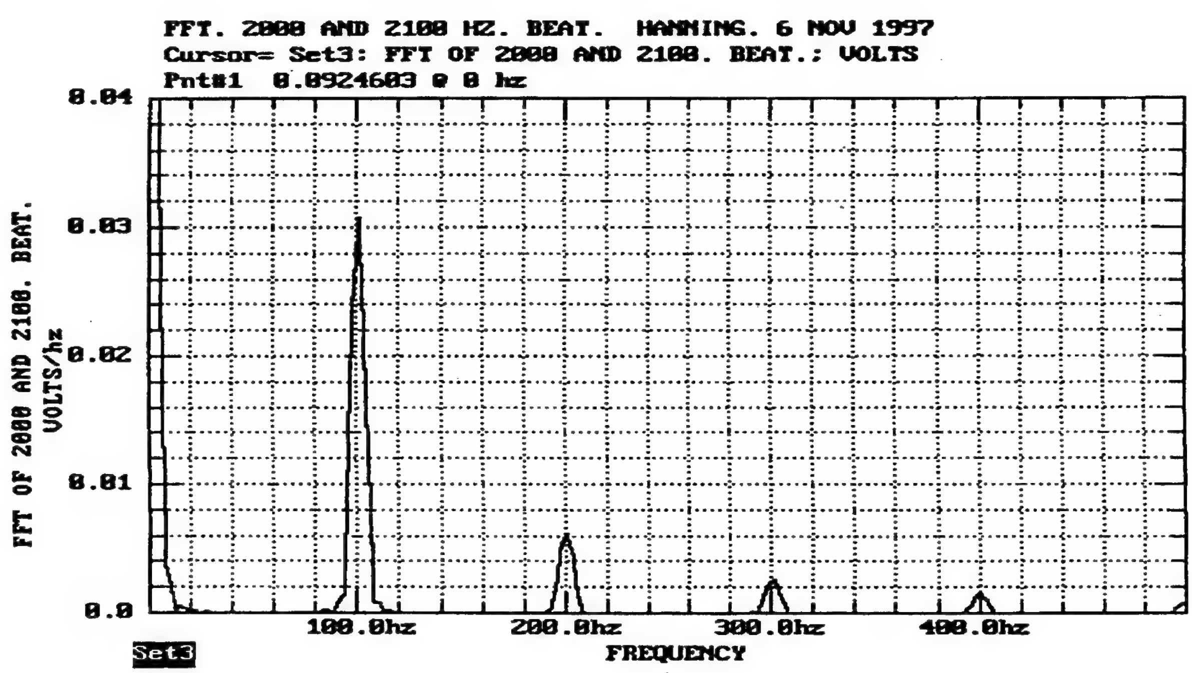

Figure 16
FFT of composite signals of 2000 and 2100 Hz

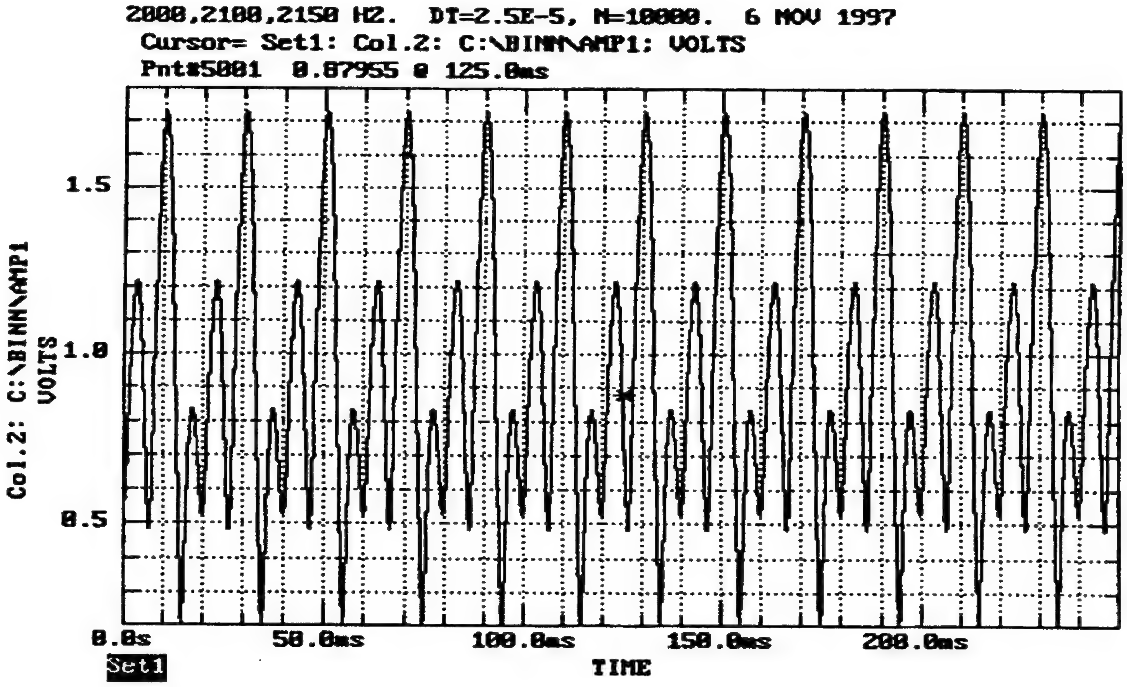

Figure 17
Composite time signal for 2000, 2100, and 2150 Hz

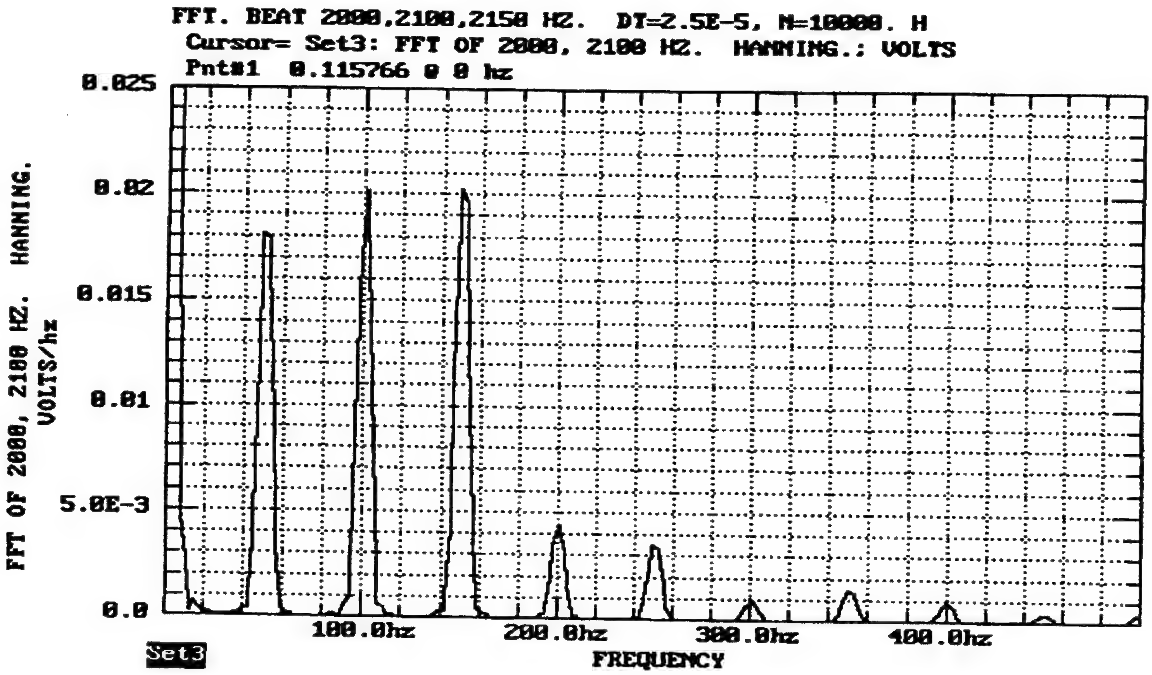

Figure 18
FFT of composite signals of 2000, 2100, and 2150 Hz

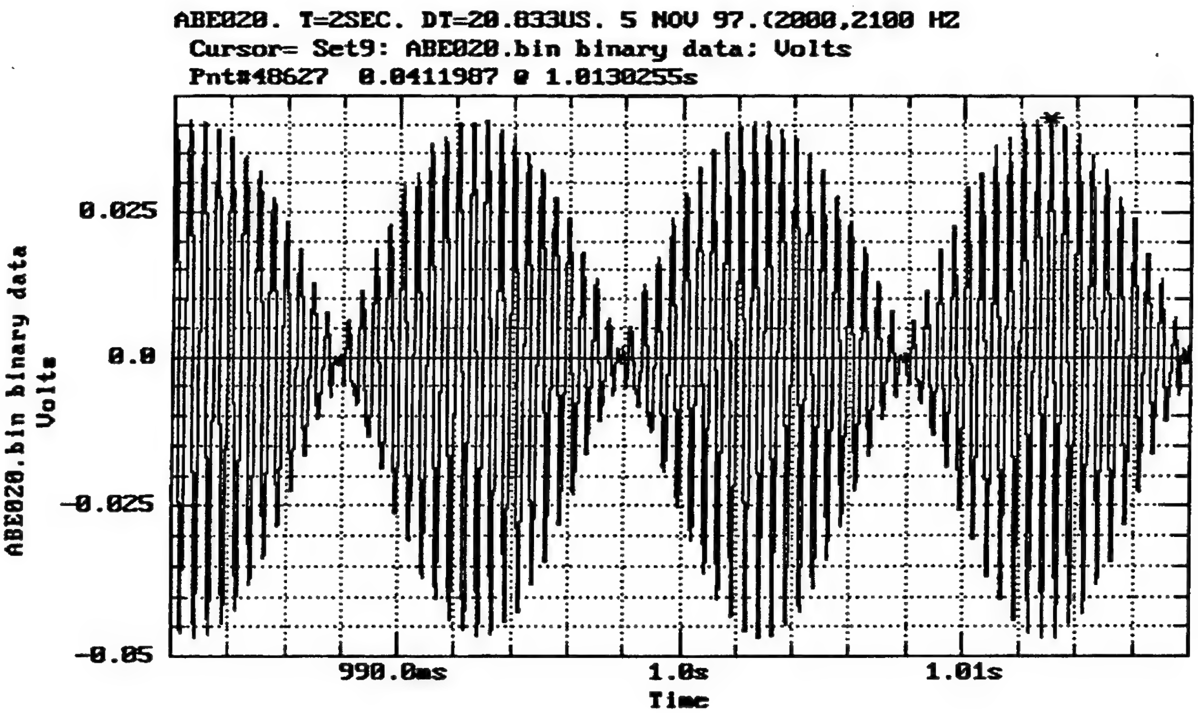

Figure 19
Experimental beat signal from 2000 and 2100 Hz

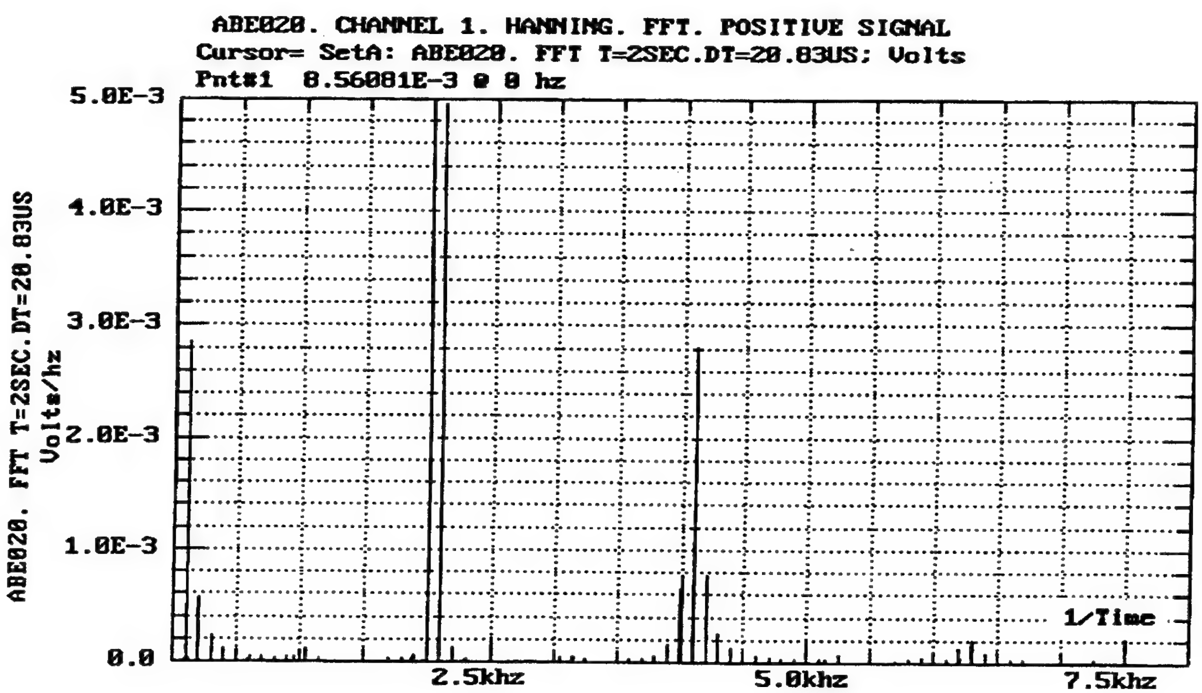

Figure 20
FFT of experimental beat signal

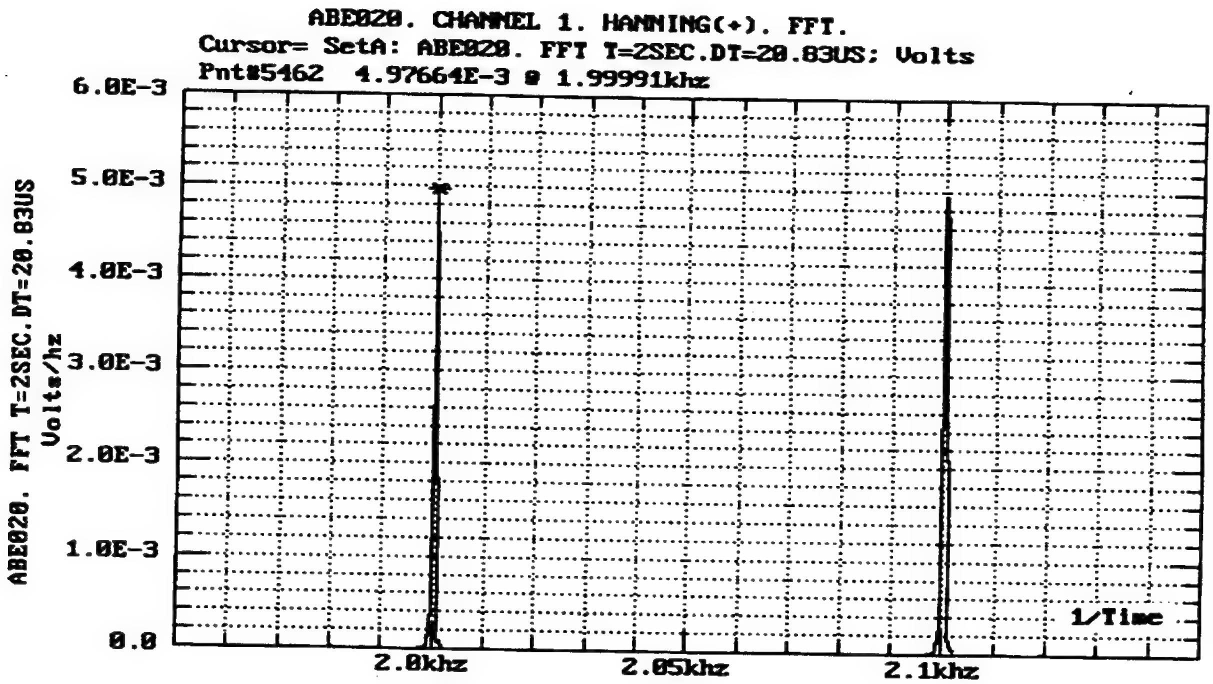

Figure 21
Detail of 2000 and 2100 Hz signal in FFT

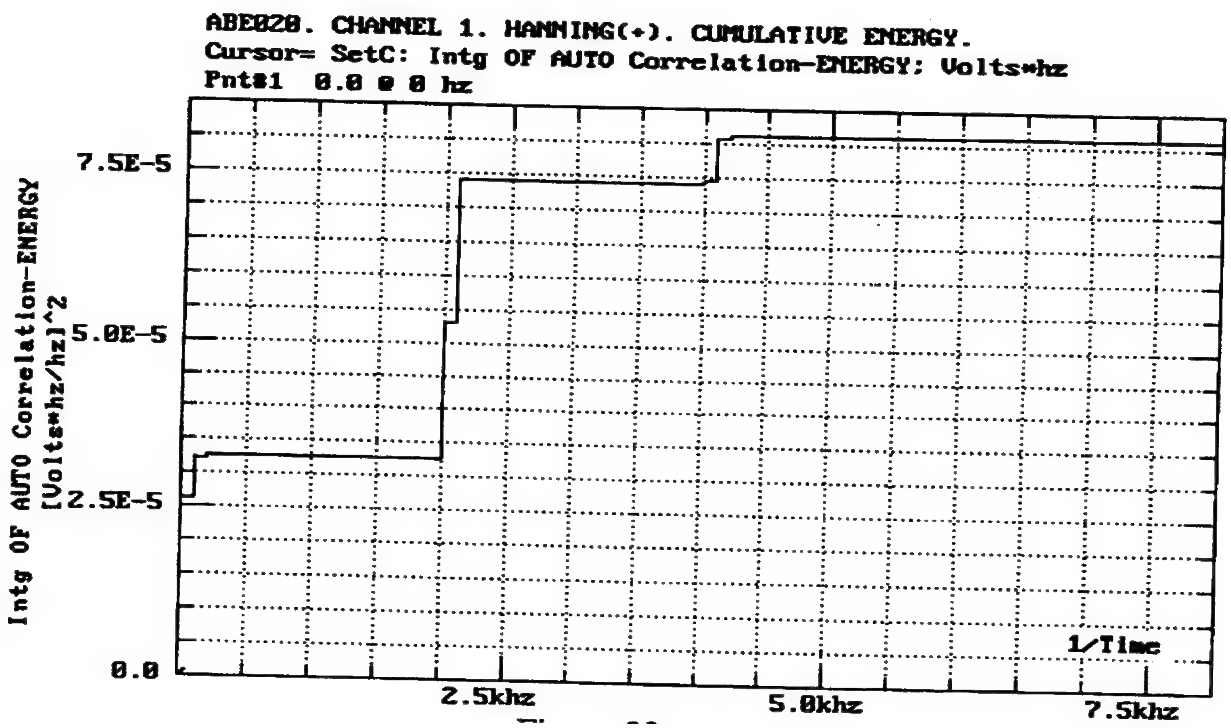

Figure 22
Cumulative energy versus frequency of experimental beat signal

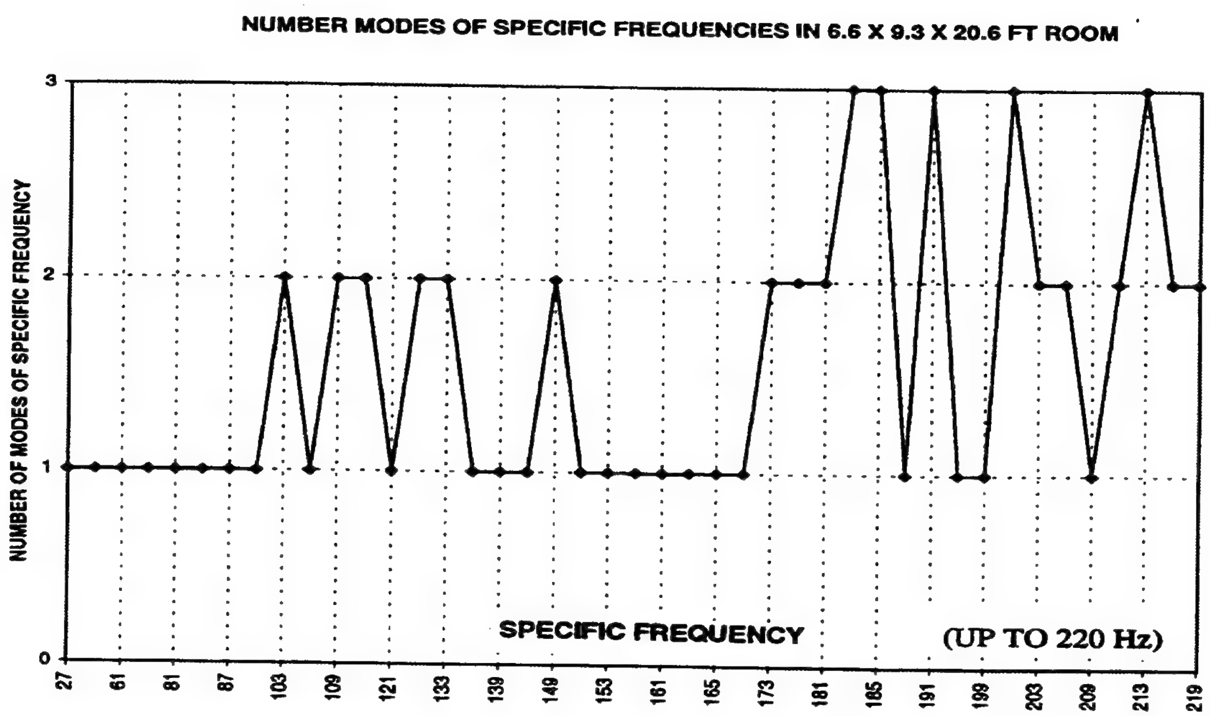

Figure 23
Number modes of specific frequencies in an enclosure

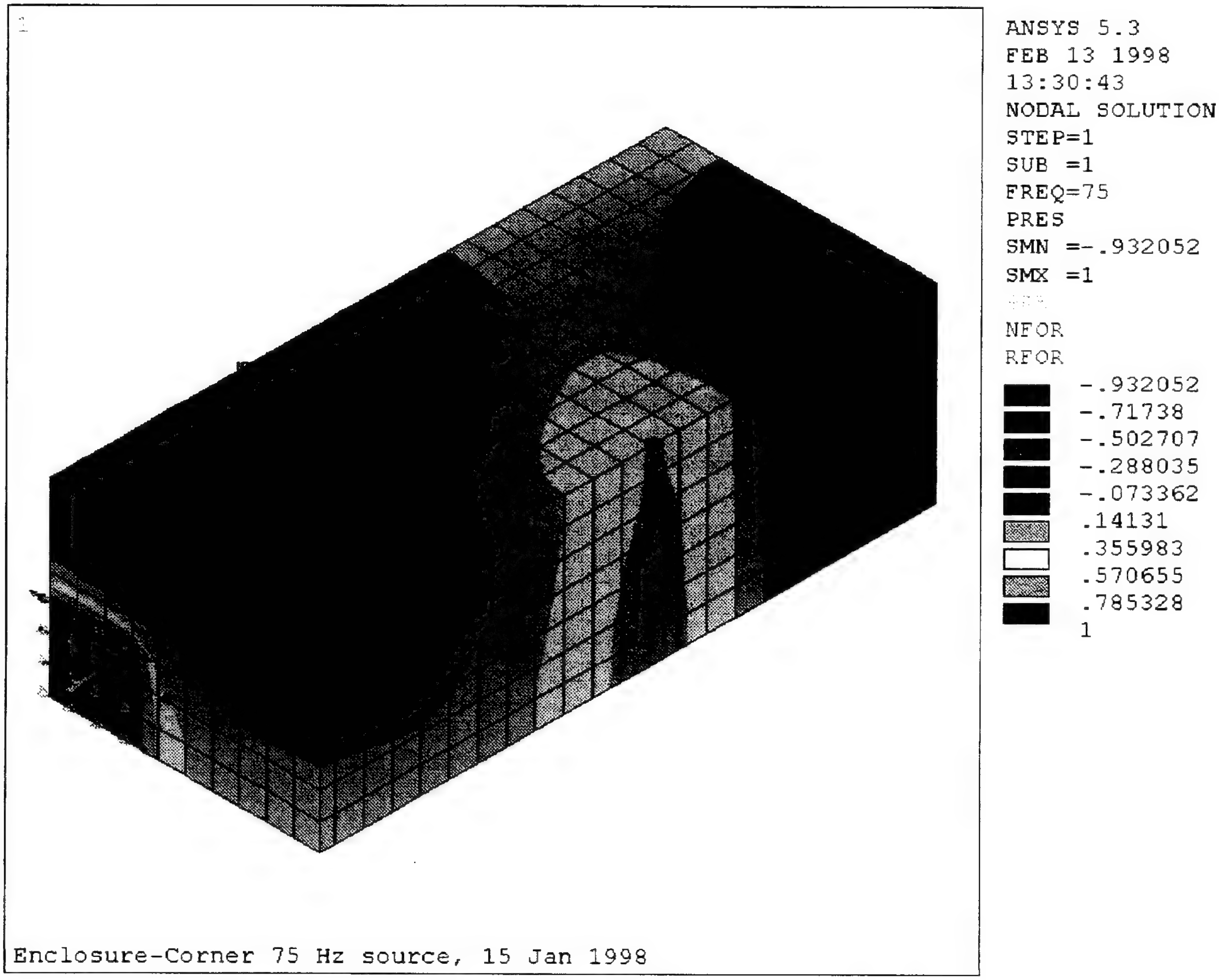

Figure 24
Finite element determined pressure modes in room driven at 75 Hz

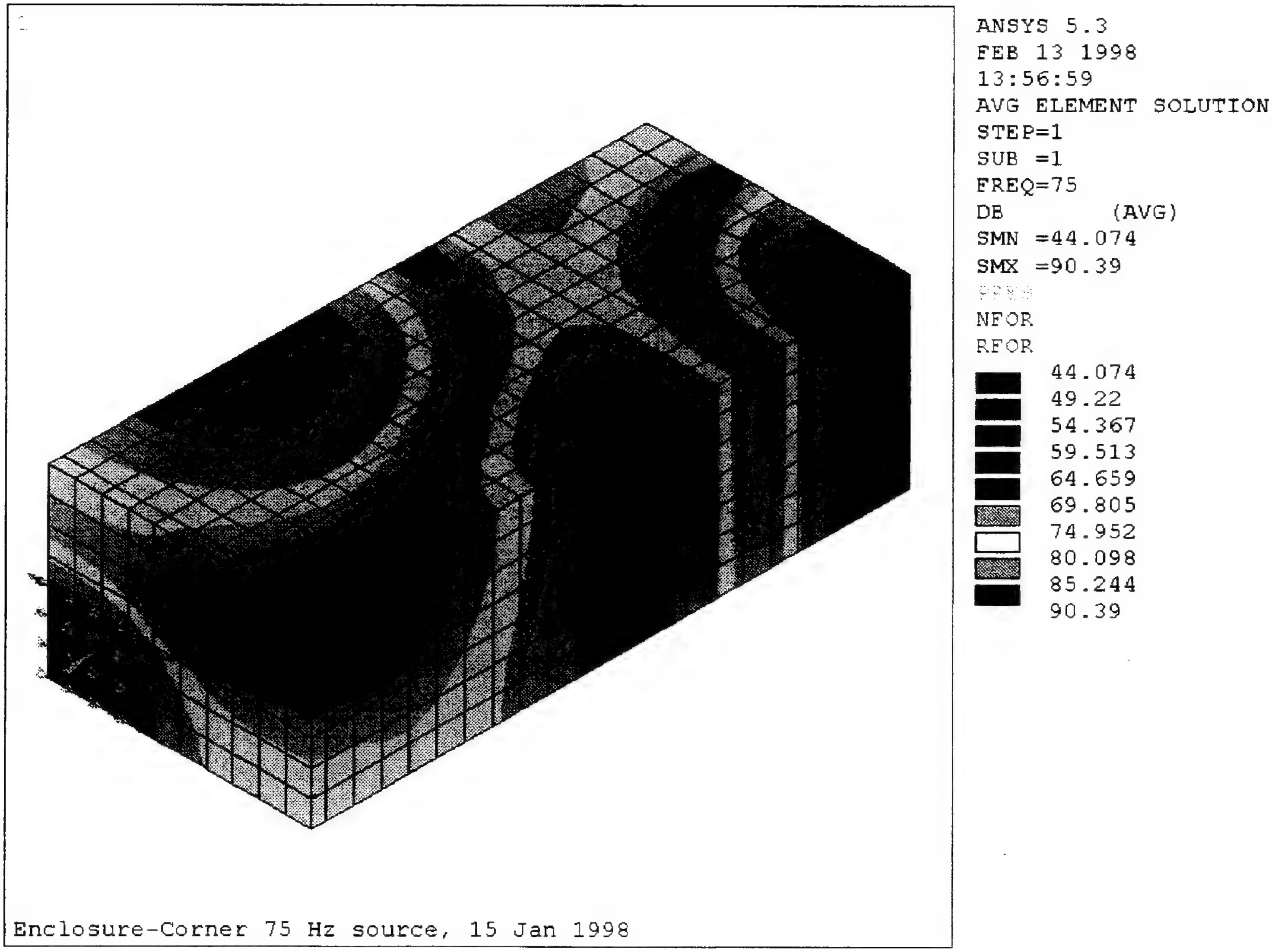

Figure 25
Decibel plot of pressure modes in room driven at 75 Hz

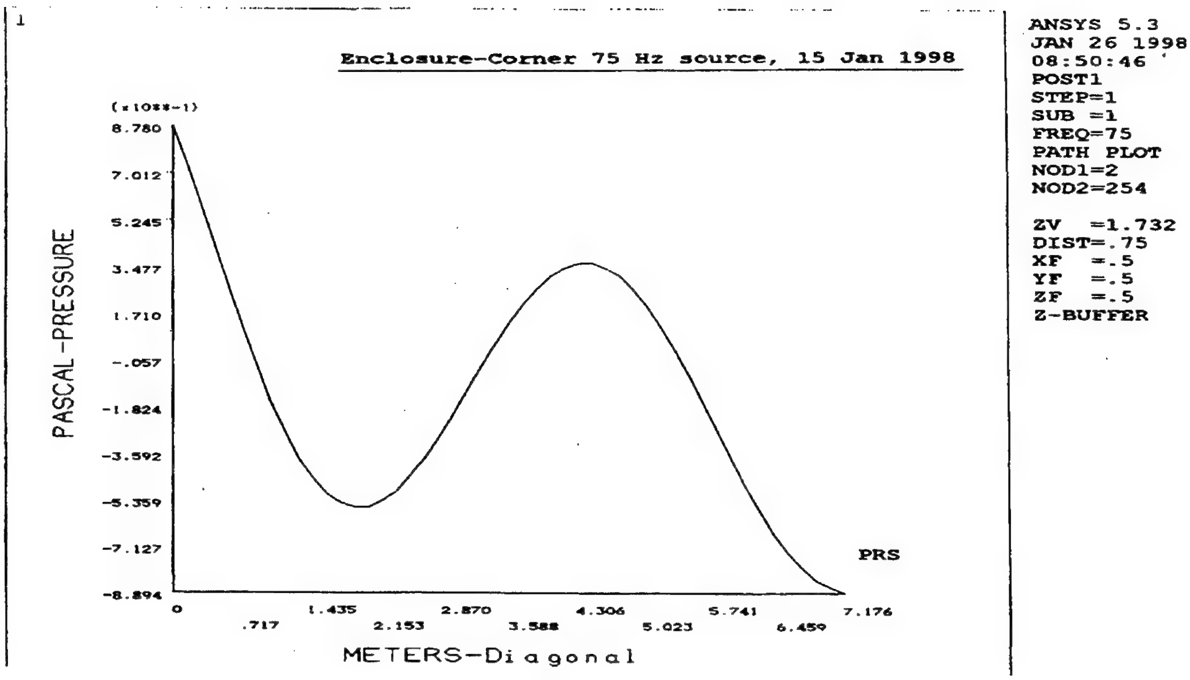

Figure 26
Pressure along diagonal in room driven at 75 Hz

Table 1
Morse complex impedance FORTRAN program

```
      PROGRAM MORSE
C ***DETERMINE COMPLEX IMPEDANCE OF MATERIALS****
C ***PHILIP MORSE, 'VIBRATION AND SOUND,' 1948, PP.239-246
C ***USE COMPLEX FUNCTION OF NATURAL EXPONENT TO
C ***OBTAIN REAL AND IMAGINARY PARTS OF IMPEDANCE. 10/24/97
C ******************************************************
C Enter 5 REAL data on one line in file MORSE separated by commas.
C Have the file MORSE in the directory, and run MORSE.EXE
C to output the REAL and IMAGINARY parts of the impedance,
C and the ordinary absorption coefficient.
C------------------------------------------------------
C 1. XLAM - wavelength in ft=sound speed/frequency: 1128/50=22.56'
C 2. XL - distance in ft between speaker and specimen face.
C   NOTE:  Speaker at x=0;  specimen face at x=L.
C 3. XA - x coordinate of first min pressure (from specimen face)
C 4. PMAX - maximum pressure (pascals or volts) found after moving
C   from XA toward the speaker.
C 5. PMIN1 - minimum pressure (pascals or volts) at location XA.
C ******************************************************
C  The program will calculate PI*ALPHA and PI*BETA and insert them
C  as the real and imaginary components of S:
C..S=PI*ALPHA - i*PI*BETA
C.. FIND ALPHA AND BETA FROM IMPEDANCE TUBE:
C ---------------------------------------
C..ALPHA FROM: TANH(PI*ALPHA) =PRESS(MIN)/PRESS(MAX)
C..BETA FROM: BETA= 1.- 2.*(XL-XA)/XLAM
C XSI=DIMENSIONLESS ACOUSTIC IMPEDANCE OF MATERIAL
C XSI= THETA - i*CHI = TANH(PI*ALPHA- i*PI*BETA)
C THETA IS THE DIMENSIONLESS MATERIAL ACOUSTIC RESISTANCE.
C CHI IS THE DIMENSIONLESS MATERIAL ACOUSTIC  REACTANCE.
C THETA = 1./MU, WHERE MU IS USED IN ANSYS PROGRAM.
C ... TAKE AS AN EXAMPLE:
C MAX PRESSURES OCCUR HALF WAVELENGTH APART.
C MIN PRESSURES OCCUR HALF WAVELENGTH APART ALSO.
C THEN, DISTANCE BETWEEN MAX AND MIN IS ABOUT QUARTER WAVELENGTH.
C......   N U M E R I C A L    E X A M P L E..........
C XA = 26 FEET AS LOCATION OF FIRST MIN
C XLAM = 22.56 FT =(C/F=1128/50).
C BETA = 1- 2.*(XL-XA)/XLAM =.64539
C PI*BETA = 2.02753
C PI*ALPHA= INVERSE TANH (PMIN1/PMAX)= ATANH(0.1/0.2)
C PI*ALPHA= 0.5*LN((1+PP)/(1-PP))
C WHERE PP = PMIN1/PMAX = 0.1/0.2 = 0.5
C PI*ALPHA = .54931
C XLAM=22.56
C XL=30.
C XA=26.
C PMAX=.2
C PMIN1=.1
```


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Table 1
(cont)

```
      COMPLEX S,XSI
      REWIND 1
      OPEN(1,FILE='MORSE')
      READ(1,111) XLAM,XL,XA,PMAX,PMIN1
111   FORMAT(5F10.5)
      WRITE(*,222) XLAM,XL,XA,PMAX,PMIN1
222   FORMAT(1X,'XLAM=',F9.4,' XL=',F9.4,' XA=',F9.4,
     1 ' PMAX=',F9.4,' PMIN1=',F9.4)
      RATIO=PMAX/PMIN1
      AC=1.-((RATIO-1.)/(RATIO+1.))**2
      PI=3.14159265
      PIBETA=PI*(1.-2.*(XL-XA)/XLAM)
C  PIALPHA IS INVERSE HYPERBOLIC TANGENT OF (PMIN1/PMAX)
      PP=PMIN1/PMAX
      PIALPHA=0.5*LOG((1.+PP)/(1.-PP))
C LOG IS TO THE BASE E.
      S=CMPLX(PIALPHA,-PIBETA)
C FORTRAN HAS COMPLEX EXPONENTIAL FUNCTION FOR TANH.
C TANH=[E(S) - E(-S)] / [E(S) + E(-S)]
C E(S) IS E TO THE POWER S.
      XSI=(CEXP(S)-CEXP(-S))/(CEXP(S)+CEXP(-S))
      WRITE(*,100) XSI
      WRITE(*,101)
      WRITE(*,102)
      WRITE(*,103)
      WRITE(*,104) AC
100   FORMAT(1X,'THETA =' F10.4,'  CHI  = ', F10.4)
101   FORMAT(1X,'THETA=DIMENSIONLESS REAL PART OF IMPEDANCE.')
102   FORMAT(1X,'  CHI=DIMENSIONLESS IMAGINARY PART OF IMPEDANCE.')
103   FORMAT(1X,'THETA=ACOUSTIC RESISTANCE, CHI=ACOUSTIC REACTANCE.')
104   FORMAT(1X,'ABSORPTION COEFFICIENT = ',F8.4)
      STOP
      END
```

Input file for 'MORSE' or 'SWEN':

22.56,30.,26.,.2,.1

Output file from 'MORSE' or 'SWEN':

```
XLAM=  22.5600 XL=  30.0000 XA=  26.0000 PMAX=    .2000 PMIN1=    .1000
THETA =    1.2630  CHI  =      .7499
THETA=DIMENSIONLESS REAL PART OF IMPEDANCE.
  CHI=DIMENSIONLESS IMAGINARY PART OF IMPEDANCE.
THETA=ACOUSTIC RESISTANCE, CHI=ACOUSTIC REACTANCE.
ABSORPTION COEFFICIENT =     .8889
Stop - Program terminated.
```

Table 2

Swenson complex impedance FORTRAN program

```
      PROGRAM SWEN
C ***DETERMINE COMPLEX IMPEDANCE OF MATERIALS
C ***GEORGE SWENSON, 'PRINCIPLES OF MODERN ACOUSTICS',1965.
C ***USE COMPLEX FUNCTION OF NATURAL EXPONENT TO
C ***OBTAIN REAL AND IMAGINARY PARTS OF IMPEDANCE. 10/24/97
C ************************************************
C Enter 5 REAL data on one line in file SWEN separated by commas.
C Have the file SWEN in the directory, and run SWEN.EXE
C to out REAL and IMAGINARY parts of the impedance,
C and the ordinary absorption coefficient.
C ------------------------------------------------
C 1. XLAM - wavelength in ft=sound speed/frequency: 1128/50=22.56'
C 2. XL - distance in ft between speaker and specimen face.
C NOTE: Speaker at x=0; specimen face at x=L.
C 3. XA - x coordinate at first min pressure from the specimen face.
C 4. PMAX - max pressure (pascals or volt) found after moving
C  from XA toward the speaker.
C 5. PMIN1 = min pessure (pascals or volt) at location XA.
C ************************************************
C Programs calculates PI*ALPHA and PI*BETA and inserts them
C as real and imaginary components of S:
C..S=PI*ALPHA - i*PI*BETA
C..FIND ALPHA AND BETA FROM IMPEDANCE TUBE.
C ********************************
C..ALPHA FROM: TANH(PI*ALPHA)= PRESS(MIN)/PRESS(MAX)
C.. BETA FROM: BETA= 1.- ((L-XA)/(XA-XB))
C..XL=LENGTH OF IMPEDANCE TUBE.
C..XA=X COORDINATE OF FIRST PRESS(MIN) FROM SPECIMEN.
C..XB=X COORDINATE OF FIRST PRESS(MAX) FROM SPECIMEN. XB<XA
C XSI=DIMENSIONLESS ACOUSTIC IMPEDANCE OF MATERIAL
C XSI= THETA - i*CHI = TANH(PI*ALPHA-i*PI*BETA)
C THETA IS THE DIMENSIONLESS MATERIAL ACOUSTIC RESISTANCE.
C CHI IS THE DIMENSIONLESS MATERIAL ACOUSTIC  REACTANCE.
C THETA = 1./MU, WHERE MU IS USED IN ANSYS PROGRAM.
C TAKE AS AN EXAMPLE:
C L=30 FEET,  XA =26.
C A= (XA-L)/XLAM
C LAMB = 22.56 FEET WAVELENGTH = (C/F = 1128 FPS/ 50 HZ).
C A= (26-30)/22.56 = -.1773
C PHI = PI*(1.- 4*A) = PI* (1-4*(-.1773)) = 5.36967
C S HAS (0.,PHI)
      COMPLEX S,XSI
      REWIND 1
      OPEN(1,FILE='SWEN')
      READ(1,111) XLAM,XL,XA,PMAX,PMIN1
111   FORMAT(5F10.5)
      WRITE(*,222) XLAM,XL,XA,PMAX,PMIN1
222   FORMAT(1X,'XLAM=',F9.4,' XL=',F9.4,' XA=',F9.4,
     1 ' PMAX=',F9.4,' PMIN1=',F9.4)
      RATIO=PMAX/PMIN1
      AC=1.-((RATIO-1.)/(RATIO+1.))**2
      XL=30.
      XA=26.
      XLAM=22.56
```

Table 2
(cont)

```
      PMAX=0.2
      PMIN1=0.1
      A=(XA-XL)/XLAM
      PHI=3.14159265*(1.-4.*A)
C CALCULATE PHI, AND PLACE THE REAL NUMBER IN SECOND T POSITION:
      S=CMPLX(0.,-PHI)
C FORTRAN HAS COMPLEX EXPONENTIAL FUNCTION FOR TANH.
C TANH=[E(S) - E(-S)] / [E(S) + E(-S)]
C E(S) IS E TO THE POWER S.
      RATIO=PMAX/PMIN1
      Q=(RATIO-1.)/(RATIO+1.)
      XSI=(1.+Q*CEXP(S))/(1.-Q*CEXP(S))
      WRITE(*,100) XSI
      WRITE(*,101)
      WRITE(*,102)
      WRITE(*,103)
      WRITE(*,104) AC
100   FORMAT(1X,'THETA =' F10.4,  ';    CHI = ', F10.4)
101   FORMAT(1X,'THETA=DIMENSIONLESS REAL PART OF IMPEDANCE.')
102   FORMAT(1X,'  CHI=DIMENSIONLESS IMAGINARY PART OF IMPEDANCE.')
103   FORMAT(1X,'THETA=ACOUSTIC RESISTANCE, CHI=ACOUSTIC REACTANCE.')
104   FORMAT(1X,'ABSORPTION COEFFICIENT =',F8.4)
      STOP
      END
```

Table 3
ASTM complex impedance FORTRAN program

```
      PROGRAM ASTM
C ASTM C384-77  IMPEDANCE AND ABSORPTION OF ACOUSTICAL
C MATERIALS BY IMPEDANCE TUBE METHOD.     (24 OCT 1997)
C *****************************************************
C Enter 7 REAL data on one line in file ASTM separated by commas.
C Have file ASTM in directory, and run ASTM.EXE
C to output REAL and IMAGINARY parts of impedance,
C and the ordinary absorption coefficient.
C -----------------------------------------------------
C 1. XLAM - wavelength in feet=sound speed/frequency: 1128/50=22.56'
C 2. XAC - x coordinate of first MIN pressure (from end plate face)
C   during calibration (metal end only in place; NO SPECIMEN).
C 3. XCC - x coordinate of second MIN pressure (going toward speaker)
C   during calibration (metal end only in place; NO SPECIMEN).
C 4. XA - x coordinate of first MIN pressure (from specimen face).
C 5. PMAX - MAX pressure (pascal or volt) between the two MINS.
C 6. PMIN1 - MIN pressure at XA (location of 1st min: SPECIMEN)
C 7. PMIN2 - MIN pressure at XC (location of 2nd min; SPECIMEN)
C *********** N U M E R I C A L    E X A M P L E ********
C STEP 1:  CALIBRATION. METAL PLATE AT CLOSURE END X=XL=30 FT.
C XA=FIRST MIN (NO SPECIMEN) = XL-LAM/4     = 30-5.64 =     24.36
C XC=SECOND MIN (NO SPECIMEN)= XAA - XLAM/2 = 24.36-11.28 = 13.08
C      XAC=24.36
C      XCC=13.08
C      XL=(3.*XAC-XCC)/2.
C STEP 2: SPECIMEN FACE IN POSITION AT X=XL:
C REMEASURE FIRST AND SECOND MIN PRESSURES: (LOCATION XC NOT NEEDED)
C      XA=26.
C      XC=14.72
C      D1=XL-XA
C D2 IS ABOUT HALF WAVELENGTH.... D2=XLAM/2 = 11.28
C      D2=11.28
C PMIN1,PMAX,PMIN2 ARE PRESSURES AT XA,XB,XC:
C THESE UNITS CAN BE P A S C A L S   OR   V O L T S:
C PMAX= .2, PMIN1 = .1,  PMIN2= .09
      REWIND 1
      OPEN(1,FILE='ASTM')
      READ(1,110) XLAM,XAC,XCC,XA,PMAX,PMIN1,PMIN2
110   FORMAT(7F8.3)
      WRITE(*,111) XLAM,XAC,XCC,XA
      WRITE(*,112) PMAX,PMIN1,PMIN2
111   FORMAT(1X,'XLAM,XAC,XCC,XA = ',5(1X,F8.3))
112   FORMAT(1X,'PMAX,PMIN1,PMIN2= ',3(1X,F8.3))
      XL=(3.*XAC-XCC)/2.
      D1=XL-XA
      D2=XLAM/2.
C USE FORMULAS 11,13 IN ASTM FOR ABSORPION COEFFICIENT:
      RATIO1=PMAX/PMIN1
      RATIO2=PMAX/PMIN2
      AB1=1.-((RATIO1-1.)/(RATIO1+1.))**2
      AB2=1.-((RATIO2-1.)/(RATIO2+1.))**2
      AB11=AB1-(AB2-AB1)/2.
      AB13=AB1-(D1/D2)*(AB2-AB1)
```

Table 3
(cont)

```
C CONVERT PRESSURE UNITS TO DB:
      PMAX=20.*LOG10(PMAX/29E-6)
      PMIN1=20.*LOG10(PMIN1/29E-6)
      PMIN2=20.*LOG10(PMIN2/29E-6)
      XL1=PMAX-PMIN1
      XL2=PMAX-PMIN2
      XL0=XL1+(XL1-XL2)/2.
C XK0 = 10**(XL0/20)
      XK0=EXP((XL0/20)*ALOG(10.))
      XM=(XK0+(1./XK0))/2.
      XN=(XK0-(1./XK0))/2.
      PHI=(D1/D2 -0.5)
      PHID=360.*PHI
      THETA=1./(XM-XN*COS(PHI))
      CHI= (XN*SIN(PHI))/(XM-XN*COS(PHI))
      WRITE(*,100) XL1,XL2,XL0,XK0
      WRITE(*,200) D1,D2,XM,XN,PHI
      WRITE(*,300) THETA,CHI
      WRITE(*,400) AB11,AB13
100   FORMAT(1X,'XL1,2,0,K0      =',4(1X,F9.5))
200   FORMAT(1X,'D1,D2,M,N,PHI  =',5(1X,F9.5))
300   FORMAT(1X,'THETA, CHI = ', 2F12.4)
400   FORMAT(1X,'ABSORPTION COEFFICIENT (EQ.11,13)=',2(1X,F7.4))
      STOP
      END
```

Input file for 'ASTM':

22.56,24.36,13.08,26.,.2,.1,.09

Output file from 'ASTM":

```
XLAM,XAC,XCC,XA =     22.560    24.360    13.080    26.000
PMAX,PMIN1,PMIN2=       .200      .100      .090
XL1,2,0,K0      =   6.02060   6.93575   5.56302   1.89737
D1,D2,M,N,PHI   =   4.00000  11.28000   1.21221    .68516   -.14539
THETA, CHI =        1.8717     -.1858
ABSORPTION COEFFICIENT (EQ.11,13)=   .9053   .9005
Stop - Program terminated.
```

Table 5
Multiple array source beat frequency FORTRAN program

```
      PROGRAM BEAT
C BEAT FREQUENCY FOR MULTIPLE POINT ACOUSTIC SOURCES FOR OPEN REGION.
C FOR RESPONSE FOR UP TO THREE MICROPHONE LOCATIONS.
C INPUT FILE IS ARRANGED AS FOLLOWS:(COMMA SEPARATION BETWEEN DATA)
C 1.NUMBER OF SPEAKER SOURCES(MAX=20); NUMBER MIKES(MAX=3)
C 2.THE FREQUENCIES OF THE SOURCES (IN ORDER)
C 3.THE X COORDINATE (LATERAL) OF THE SOURCES (ALL IN X,Z PLANE).
C 4.THE Z COORDINATE (VERTICAL) OF THE SOURCES (ALL IN X,Z PLANE)
C 5.THE A GEOMETRIC REGRESSION COEFFICIENT FOR THE SOURCES.
C 6.THE B GEOMETRIC REGRESSION COEFFICIENT FOR THE SOURCES (S=A*DIST**B)
C 7.THE XM COORDINATE (LATERAL) OF THE MIKES.
C 8.THE YM COORDINATE (AXIS) OF THE MIKES. (SOURCES ARE AT YM=0.)
C 9.THE ZM COORDINATE (VERTICAL) OF THE MIKES.
C 10.NUMBER OF CYCLES OF FIRST FREQUENCY, AND DATA POINTS/CYCLE.
C 11.SPEED OF SOUND.
C++NOTE: IF SOUND SPEED IN  M/SEC, ALL DISTANCES IN METERS.(344 M/S)
C++NOTE: IF SOUND SPEED IN FT/SEC, ALL DISTANCES IN FEET. (1128 FPS)
C....IF A FREQUENCY IS A FUNCTION OF TIME, IT MUST BE PROGRAMMED.
C...EXAMPLE OF THE SOURCE FILE WITH THREE SOURCES, TWO MIKES:
C  3,2              :THREE SOURCES, TWO MIKES
C  2000,2100,2150 :FREQUENCIES OF THE THREE SOURCES.
C  0,0,0            :LATERAL X COORDINATE OF SOURCES 1,2,3.
C  0,4,8            :VERTICAL Z COORDINATE OF SURCES 1,2,3.
C  110.,150.,180. :A COEFFICIENT FOR SOURCES 1,2,3
C  -1.13,-1.17,-1.22 :B EXPONENT FOR SOURCES 1,2,3
C  0.,4.               :LATERAL XM COORDINATE FOR MIKES 1,2
C  100.,50.            :AXIS YM COORDINATE FOR MIKES 1,2
C  2.,2.               :VERTICAL ZM COORDINATE FOR MIKES 1,2
C  500.,20.             :NUMBER CYCLES OF 2000 HZ WITH 20 DATA/CYCLE
C  1128.               :SPEED OF SOUND (HERE 1128 FT/SEC FOR 20 C)
C---- PLOT AMP1,AMP2,AMP3 FOR RESULTANT AMPLITUDE AT MIKES 1,2,3
C---- PLOT PHA1,PHA2,PHA3 FOR RESULTANT PHASE (NOT AS SIGNIFICANT).
      DIMENSION F(20),X(20),Z(20),A(20),B(20),XM(3),YM(3),ZM(3)
      DIMENSION R(20),S(20),DELX(20),DELZ(20)
      REWIND 10
      OPEN(10,FILE='BEAT')
      READ(10,111) NSOURCE,NMIKE
      READ(10,112) (F(I),I=1,NSOURCE)
      READ(10,112) (X(I),I=1,NSOURCE)
      READ(10,112) (Z(I),I=1,NSOURCE)
      READ(10,112) (A(I),I=1,NSOURCE)
      READ(10,112) (B(I),I=1,NSOURCE)
      READ(10,112)(XM(I),I=1,NMIKE)
      READ(10,112)(YM(I),I=1,NMIKE)
      READ(10,112)(ZM(I),I=1,NMIKE)
      READ(10,112) CY,DCY
      READ(10,112) C
111   FORMAT(2I3)
112   FORMAT(10F8.3)
113   FORMAT(10F8.2)
      WRITE(*,111) NSOURCE,NMIKE
      WRITE(*,113) (F(I),I=1,NSOURCE)
      WRITE(*,112) (X(I),I=1,NSOURCE)
      WRITE(*,112) (Z(I),I=1,NSOURCE)
```

Table 5
(cont)

```
      WRITE(*,112) (A(I),I=1,NSOURCE)
      WRITE(*,112) (B(I),I=1,NSOURCE)
      WRITE(*,112) (XM(I),I=1,NMIKE)
      WRITE(*,112) (YM(I),I=1,NMIKE)
      WRITE(*,112) (ZM(I),I=1,NMIKE)
      WRITE(*,112) CY,DCY,C
      REWIND 1
      REWIND 2
      REWIND 3
      REWIND 4
      REWIND 7
      REWIND 8
      OPEN(1,FILE='AMP1')
      OPEN(2,FILE='AMP2')
      OPEN(3,FILE='AMP3')
      OPEN(4,FILE='PHA1')
      OPEN(7,FILE='PHA2')
      OPEN(8,FILE='PHA3')
      PI=3.141592654
C NUMBER OF MICROPHONES IN FIELD
      DO 1 I=1,NMIKE
C  CY=NUMBER OF CYCLES OF FIRST FREQUENCY F(1) TO BE CONSIDERED =500.
C  DCY=POINTS PER CYCLE FOR DIGITIZING = 20
C..DT=SAMPLING TIME = 1/(F(1)*DCY) = 1/(2000*20) =2.5E-5
C.. T=TOTAL TIME = CY/F(1) = 500/2000 = .25 SEC
C.. N=NUMBER DATA = T/DT = .25/2.5E-5 = 10000
      N=CY*DCY
      DT=1./(F(1)*DCY)
      WRITE(*,400)
400   FORMAT(15X,'SIGNAL STRENGTH   DISTANCE R BETWEEN',/
     1 5X,'MIKE  SOURCE   S AT R        SOURCE AND MIKE')
C FIND DISTANCE OF THE SOURCES AND SOURCE PEAK TO THIS MIKE:
      DO 3 J=1,NSOURCE
      DELX(J)=XM(I)-X(J)
      DELZ(J)=ZM(I)-Z(J)
      RS=SQRT(YM(I)*YM(I)+DELZ(J)*DELZ(J))
      R(J)=SQRT(RS*RS+DELX(J)*DELX(J))
C CALCULATE SIGNAL SOURCE AT THE MIKE FROM GEOMETRIC REGRESSION A,B:
      S(J)=A(I)*(R(J))**B(I)
      WRITE(*,401) I,J,S(J),R(J)
401   FORMAT(2X,2I6,2(3X,F10.3))
3     CONTINUE
C DO TIME CALCULATIONS
      DO 2 K=1,N
      T=DT*(K-1)
C SUM UP ALL THE CONTRIBUTIONS OF SIGNAL SOURCES
      SUMREAL=0.
      SUMIMAG=0.
      DO 22 KK=1,NSOURCE
      SREAL=S(KK)*COS(2.*PI*F(KK)*(T+R(KK)/C))
      SIMAG=S(KK)*SIN(2.*PI*F(KK)*(T+R(KK)/C))
      SUMREAL=SUMREAL+SREAL
      SUMIMAG=SUMIMAG+SIMAG
22    CONTINUE
```

Table 5
(cont)

```
C FIND AMPLITUDE AND PHASE OF COMPOSITE SIGNAL AT MIKE (I):
      IF(SUMIMAG.EQ.0.) SUMIMAG=1E-8
      AMP=SQRT(SUMREAL*SUMREAL+SUMIMAG*SUMIMAG)
      PHA=180.*2.*ATAN(SUMREAL/SUMIMAG)/(PI*2.)
      IF(I.EQ.1)WRITE(1,500) T,AMP
      IF(I.EQ.2)WRITE(2,500) T,AMP
      IF(I.EQ.3)WRITE(3,500) T,AMP
      IF(I.EQ.1)WRITE(4,500) T,PHA
      IF(I.EQ.2)WRITE(7,500) T,PHA
      IF(I.EQ.3)WRITE(8,500) T,PHA
500   FORMAT(2F10.5)
2     CONTINUE
      WRITE(*,402)
402   FORMAT(1X,50(1H*))
1     CONTINUE
      WRITE(*,403) N,DT
403   FORMAT(1X,'N,DT =',I6,1X,F12.6)
      STOP
      END
```

Input file for 'BEAT':

```
3,2
2000.,2100.,2150.
0.,0.,0.
0.,4.,8.
110.,150.,180.
-1.13,-1.17,-1.22
20.,2.
100.,50.
20.,12.
500.,20.
1128.
```

Output file from 'BEAT":

```
   .000     .000     .000
   .000    4.000    8.000
110.000  150.000  180.000
 -1.130   -1.170   -1.220
 20.000    2.000
100.000   50.000
 20.000   12.000
500.000   20.0001128.000
                 SIGNAL STRENGTH   DISTANCE R BETWEEN
    MIKE   SOURCE     S AT R       SOURCE AND MIKE
     1       1          .579       103.923
     1       2          .583       103.228
     1       3          .587       102.684
**************************************************
                 SIGNAL STRENGTH   DISTANCE R BETWEEN
    MIKE   SOURCE     S AT R       SOURCE AND MIKE
     2       1         1.492        51.459
     2       2         1.519        50.675
     2       3         1.536        50.200
**************************************************
N,DT = 10000        .000025
Stop - Program terminated.
```

Table 6

Frequency modes in closed room FORTRAN program

```
      PROGRAM MODES
C MODES IN SHOEBOX ROOM. D(X),D(Y),D(Z)
C ROOM IN SIZES UP TO (X,Y,Z) IN INCREMENTS IN EACH.
C DO FOR SINGLE SIZE: X=9.32, Y=6.562, Z=20.6 FT
C SOUND SPEED C TAKEN AS 1128 FT/SEC.
C FIND F(I,J,K) FREQUENCIES, AND ASSOC WAVELENGTH W(I,J,K)
C W(X)=C/F(X)          W(I)=C/F(I)
C FOR EACH ROOM CONDITION, INTEGERS NX,NY,NZ VARY FROM 1 ON.
C FOR CLOSED WALLS: W(X)=2*D(X)/NX
C FOR EACH DIMENSION D, INTEGERS GO FROM 1,2,3,,,,
C CONDITIONS:
C W(I).LE.2*D(I)/N(I)       OR: W(X).LE.2*D(X)/NX
C [IF ONE END CLOSED, OTHER OPEN: W(I).LE.4*D(I)/N(I) ]
C ...METHOD:  FOR FIXED SIZE ROOM, VARY INTEGERS FOR
C SIMPLE FREQENCIES IN UNCLUTTERED ROOM..
C I,J,K  = X,Y,Z DIRECTIONS...... DISTANCES IN FEET.
      DIMENSION DX(50),DY(50),DZ(50),NX(50),NY(50),NZ(50),
     1 W(38,38,38),F(38,38,38),A(1000),B(1000)
      REWIND 1
      REWIND 2
      REWIND 3
      OPEN(1,FILE='FREQ')
      OPEN(2,FILE='WAVE')
      OPEN(3,FILE='FREQN')
C FILE FREQ GIVES FREQUENCIES ALLOWED, AND MODES AT EACH FREQUENCY
C FILE FREQN GIVES FREQUENCIES ALLOWED, AND ACCUMULATED NUMBER..
C FIND NUMBER MODE IN 2 HZ INTERVALS, UP TO IA*2 HERTZ (IA GROUPS)
C IA MAXIMUM = 1000
      IA=110
      DO 77 I=1,IA
      A(I)=0.
      B(I)=0.
77    CONTINUE
      FMIN=100000.
      C=1128.
      C2=C/2.
      INDEX=10
      DO 1 K=1,1
C     DZ(K)=K
      DZ(K)=20.6
      KS=K
      DO 2 J=1,1
C     DY(J)=J
      DY(J)=9.32
      JS=J
      DO 3 I=1,1
C     DX(I)=I
      DX(I)=6.562
      IS=I
C VARY THE INTEGERS..
      NUM=38
      DO 4 KK=1,NUM
      NZ(KK)=KK-1
      DO 5 JJ=1,NUM
      NY(JJ)=JJ-1
```

Table 6
(cont)

```
      DO 6 II=1,NUM
      NX(II)=II-1
      FF=C2*(SQRT((NX(II)/DX(I))**2 + (NY(JJ)/DY(J))**2
     1          + (NZ(KK)/DZ(K))**2))
      IF(FF.GT.0.) WW=2./(SQRT((NX(II)/DX(I))**2 + (NY(JJ)/DY(J))**2
     1          + (NZ(KK)/DZ(K))**2))
C STORE AN ACCEPTABLE STANDING WAVELENGTH
      W(II,JJ,KK)=WW
      F(II,JJ,KK)=FF
200   FORMAT(F10.2)
C FIND CONDITIONS WHEN LOWEST FREQUENCY OCCURS..
      IF(FF.LT.FMIN) GO TO 50
      GO TO 51
50    XL=DX(I)
      YL=DY(J)
      ZL=DZ(K)
      IXN=II-1
      JYN=JJ-1
      KZN=KK-1
      FMIN=FF
51    CONTINUE
100   FORMAT(1X,'ROOM, K#,  WAVEL, FREQ= ',3I3,2X,3I3,2(1X,F8.2))
      WRITE(2,600) WW,FF
      DELF=2.
      DO 300 IN=1,IA
      DFB=DELF*IN
      DFA=DELF*(IN-1)
C A GIVES THE FREQUENCY(MEAN OF 1 HZ), B GIVES NUMBER MODES.
      IF(FF.GT.DFA.AND.FF.LE.DFB) A(IN)=(DFB+DFA)/2.
      IF(FF.GT.DFA.AND.FF.LE.DFB) B(IN)=B(IN)+1.
300   CONTINUE
10    CONTINUE
6     CONTINUE
5     CONTINUE
4     CONTINUE
3     CONTINUE
2     CONTINUE
1     CONTINUE
      BMODES=0
9     DO 90 IN=1,IA
C DO NOT WRITE IF NO FREQUENCIES WERE FOUND:
      IF(B(IN).EQ.0.) GO TO 90
      WRITE(1,600) A(IN),B(IN)
600   FORMAT(2F10.2)
      BMODES=BMODES+B(IN)
      WRITE(3,600) A(IN),BMODES
      WRITE(*,400) B(IN),BMODES,A(IN)
400   FORMAT(1X,'FREQ INTERVAL, NUMBER IN, TOT MOD= ',3(1X,F10.2))
90    CONTINUE
      WRITE(*,222) XL,YL,ZL,IXN,JYN,KZN
222   FORMAT(1X,'ROOM SIZE;INDEX FOR LOWEST FREQ= ', 3F6.2,2X,3I4)
      STOP
      END
```

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1. Philip Morse, 'Vibration and Sound,' 1948, pp. 239-246.

2. Swenson, 'Principles of Modern Acoustics,' 1965, pp.86-95.

3. ASTM, C384-90A, Impedance and Absorption of Acoustic Materials.

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